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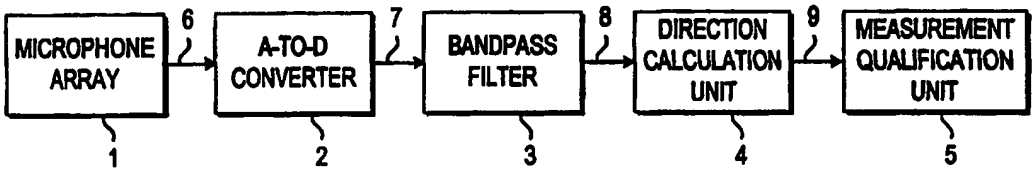

(57) Abstract

A system for wave source direction finding using sensor arrays (1), including an approximate direction finder (21), a precise direction finder (22), and a measurement qualification unit (5) for evaluating the validity of the precise direction.

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WAVE SOURCE DIRECTION DETERMINATION WITH SENSOR ARRAY

1. FIELD OF THE INVENTION

The present invention relates generally to signal processing, and more specifically to a system and method using signal processing for finding the direction of a particular wave source using an array of sensors. All documents cited herein, and all documents referenced in documents cited herein, are hereby incorporated by reference.

2. BACKGROUND OF THE INVENTION

A system for finding or tracking the direction of a particular wave source has many applications. One example is a directional microphone system, where a microphone is to be pointed to the direction of a particular sound source. Another is a video conferencing system, where a camera needs to be moved to the direction of the participating speaker.

One well-known technique of wave-source direction finding is beamforming. Beamforming, itself well-known in the art, uses an array of sensors located at different points in space. Connected to the array of sensors is a spatial filter that combines the signals received from the sensors in a particular way so as to either enhance or suppress signals coming from certain directions relative to signals from other directions.

Where the sensors are microphones, unless two microphones are located at equidistant from a sound source (i.e., arranged so that the line connecting the two microphones is perpendicular to the direction of the sound source), sound originating from the sound source arrives at any two microphones at different times, thereby producing a phase difference in the received signals.

If the received signals are appropriately delayed and combined by changing the spatial filter coefficients, the behavior of the microphone array can be adjusted such that it exhibits maximum receiving sensitivity toward a particular direction. In other words, the direction of maximum receiving sensitivity (so called "looking direction" of the microphone array) can be steered without physically changing the direction of the microphone array. It is then possible to determine the

direction of a particular sound source by logically (computationally) steering the looking direction of the microphone array over all directional angles and looking for the angle that produces the maximum signal strength.

However, the use of beamforming for sound-source direction finding has several drawbacks. First, a typical beamforming profile of receiving sensitivity over the angles of looking direction is so flat that it is, as a practical matter, difficult to find the peak point of maximum signal strength unless an inconvenience of a large microphone array is used. For example, the 3-dB attenuation points (reference points for signal discrimination) of a typical 15-cm microphone array may be separated by as much as 100 degrees. At small angles such as 5 degrees, the corresponding attenuation is insignificant. As a result, even a slight numerical error or noise may perturb the result, giving an erroneous direction.

Second, beamforming involves scanning the space for the direction producing the maximum received signal strength. Finding the source direction in terms of a horizonal direction (azimuth) and a vertical direction (elevation) involves searching two-dimensional space, which is computationally expensive.

Third, in order to determine the source direction with a high spatial accuracy, it is necessary to perform the beamforming calculation at a very high resolution (for example, every 1 degree). This requires delaying and summing the received signals at a very small delay step, which, in turn, requires that the signal be sampled at a very high sampling rate, imposing a severe computational burden.

Another method of finding the direction of a sound source is to measure time delays between a pair of sensors. For example, Hong Wang & Peter Chu, Voice Source Localization for Automatic Camera Pointing System in Videoconferencing, Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing, April 1997, pp. 187-90, disclose an array of microphones mounted on a vertical plane, three of them arranged in a horizonal line, and the fourth located above the center one of the three. The horizontal direction of a sound source (azimuth) is calculated by measuring the time delays of incoming signals between the two remote microphones in the horizontal line. The vertical direction (elevation) is calculated by measuring the time delays of incoming signals between the center microphone in the horizontal line and the upper microphone.

The Wang & Chu system has several drawbacks. First, since all the microphones are on the same vertical plane, they produce the same delay whether sound is coming from the front or from the back. Since the system cannot distinguish between a front and a back, ambiguities are inevitable.

Second, the performance of the system is not symmetric with respect to looking sideways and looking forward. The capability of such a system to resolve and estimate the direction of a source depends on the change of time delay in response to an incremental change in the angular direction. The time delay between two incoming signals at two adjacent microphones is:

time delay = $\sin(\phi) * \text{aperture} / \text{sound-velocity}$,

where ϕ is the angle of arrival of the sound waves, measured with respect to the normal of the microphone array, and the aperture is the spacing between two nearest microphones. Note that the change in time delay is obtained as the derivative of $\sin(\phi)$, which is a function of $\cos(\phi)$. For the same incremental angular change, the resulting time delay when the looking direction is sideways (when ϕ approaches 90 degrees) is smaller than when the looking direction is forward (when ϕ approaches 0 degree). As a result, the performance of the system looking sideways is poorer than that looking forward.

Third, the Wang & Chu system does not provide any indication of how reliable the measurements used for the direction determination were. The time delay measurement between a pair of microphones may not be reliable if it was measured in the presence of noise or based on non-relevant signals. The quality of measurement would be poor if the measurement were made in a noisy environment. Also, even if the measurement were of high quality, it may not be relevant to the direction determination. For example, if the time delay measurement were a measurement of reflected sound from a wall or furniture, or sound from a repeater source such as a loud speaker connected to an audio/video conferencing system, the measurement may not even be relevant to the direction determination. The Wang & Chu system does not provide any mechanism for verifying the quality or relevancy of measurement.

Therefore, there exists a need for a system and method that can determine the direction of a wave source accurately and efficiently, and that can also indicate the quality and relevancy of the measurements on which the direction determination is based.

3. SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to provide a direction finding system capable of processing signals from an array of sensors to determine the direction of a particular wave source.

Another object of the invention is to provide a system which can estimate the wave-source direction by taking into account the directions measured from individual microphone pairs and combining them to find the best-estimated source direction.

Still another object is to provide a system which can verify the quality of measurements used to calculate the source direction and can disqualify the source direction if it is not valid under a proper measurement criterion.

These and other objects are achieved in accordance with the present invention, which is an apparatus for using an array of sensors for finding the approximate direction of the wave source in terms of the positions of a selected subset of sensors, using the approximate direction so found, determining the precise direction of the wave source, evaluating the validity of the precise direction using a measurement criterion, and disqualifying the precise direction if the measurement criterion is not met so that the measurements can be repeated. One preferred embodiment of the present invention comprises an array of analog microphones for sensing sound from a sound source, an A-to-D converter for sampling the analog signals to produce corresponding digital signals, a bandpass filter for filtering the digital signals to band-limited signals in a frequency band of interest, an approximate-direction finder for finding the approximate direction of the sound source, a precise-direction finder for finding the precise direction of the sound source based on the approximate direction, and a measurement qualification unit for verifying the validity of the precise direction using a certain measurement criterion and for disqualifying the measurement if the measurement criterion is not satisfied.

The present invention has the advantage of being computationally efficient because it does not involve a two- dimensional search of space, as a beamformer would require. It also has the advantage of performing reliably in a noisy environment because it verifies the validity of the source direction under a variety of measurement criteria and repeats the measurements if necessary.

The present invention may be understood more fully by reference to the following figures, detailed description and illustrative examples intended to exemplify non-limiting embodiments of the invention.

4. DESCRIPTION OF THE FIGURES

Fig. 1 is a functional diagram of the overall system including a microphone array, an A-to-D converter, a band-pass filter, an approximate-direction finder, a precise-direction finder, and a measurement qualification unit in accordance with the present invention.

Fig. 2 is a perspective view showing the arrangement of a particular embodiment of the microphone array of Fig. 1.

Fig. 3 is a functional diagram of an embodiment of the approximate-direction finder of Fig. 1.

Fig. 4 is a functional diagram of an embodiment of the precise-direction finder of Fig. 1.

Fig. 5 is the 3-D coordinate system used to describe the present invention.

Fig. 6A is a functional diagram of a first embodiment of the measurement qualification unit of Fig. 1.

Fig. 6B is a functional diagram of a second embodiment of the measurement qualification unit of Fig. 1.

Fig. 6C is a functional diagram of a third embodiment of the measurement qualification unit of Fig. 1.

Fig. 6D is a functional diagram of a fourth embodiment of the measurement qualification unit of Fig. 1.

Figs. 7A - 7D are a flow chart depicting the operation of a program that can be used to implement the method in accordance with the present invention.

5. DETAILED DESCRIPTION

FIG. 1 shows the functional blocks of a preferred embodiment in accordance with the present invention. The embodiment deals with finding the direction of a sound source, but the invention is not limited to such. It will be

understood to those skilled in the art that the invention can be readily used for finding the direction of other wave sources such as an electromagnetic wave source.

The system includes an array of microphones 1 that sense or measure sound from a particular sound source and that produce analog signals 7 representing the measured sound. The analog signals 7 are then sampled and converted to corresponding digital signals 8 by an analog-to-digital (A-to-D) converter 2. The digital signals 8 are filtered by a band-pass filter 3 so that the filtered signals 9 contain only the frequencies in a specific bandwidth of interest for the purpose of determining the direction of the sound source. The filtered signals 9 are then fed into an approximate-direction finder 4 which calculates an approximate direction 10 in terms of a microphone pair selected among the microphones. The precise-direction finder 5 estimates the precise-direction 11 of the sound source based on the approximate direction. The validity of the precise-direction 11 is checked by a measurement qualification unit 6, which invalidates the precise direction if it does not satisfy a set of measurement criteria. Each functional block is explained in more detail below.

5.1. Microphone Array

FIG. 2 shows an example of the array of microphones 1 that may used in accordance with the present invention. The microphones sense or measure the incident sound waves from a sound source and generate electronic signals (analog signals) representing the sound. The microphones may be omni, cardioid, or dipole microphones, or any combinations of such microphones.

The example shows a cylindrical structure 21 with six microphones 22-27 mounted around its periphery, and an upper, center microphone 28 mounted at the center of the upper surface of the structure. The upper, center microphone is optional, but its presence improves the accuracy of the precise direction, especially the elevation angle. Although the example shows the subset of microphones in a circular arrangement of the microphone array, the microphone array may take on a variety of different geometries such as a linear array or a rectangular array.

5.2 A-to-D Converter

The analog signals representing the sound sensed or measured by the microphones are converted to digital signals by the A-to-D converter 2, which samples the analog signals at an appropriate sampling frequency. The converter may employ a well-known technique of sigma-delta sampling, which consists of oversampling and built-in low-pass filtering followed by decimation to avoid aliasing, a phenomenon due to inadequate sampling.

When an analog signal is sampled, the sampling process creates a mirror representation of the original frequencies of the analog signal around the frequencies that are multiples of the sampling frequency. "Aliasing" refers to the situation where the analog signal contains information at frequencies above one half of the sampling frequency so that the reflected frequencies cross over the original frequencies, thereby distorting the original signal. In order to avoid aliasing, an analog signal should be sampled at a rate at least twice its maximum frequency component, known as the Nyquist frequency.

In practice, a sampling frequency far greater than the Nyquist frequency is used to avoid aliasing problems with system noise and less-than-ideal filter responses. This oversampling is followed by low-pass filtering to cut off the frequency components above the maximum frequency component of the original analog signal. Once the digital signal is Nyquist limited, the rate must be reduced by decimation. If the oversampling frequency is n times the Nyquist frequency, the rate of the digital signal after oversampling needs to be reduced by decimation, which takes one sample for every n samples input.

An alternative approach to avoid aliasing is to limit the bandwidth of signals using an analog filter that halves the sampling frequency before the sampling process. This approach, however, would require an analog filter with a very sharp frequency cut-off characteristic.

5.3 Bandpass Filter

The purpose of the bandpass filter 3 is to filter the signals sensed or measured by the microphones so that the filtered signals contain those frequencies optimal for detecting or determining the direction of the signals. Signals of too low a frequency do not produce enough phase difference at the microphones to accurately detect the direction. Signals of too high a frequency have less signal energy and are thus more subject to noise. By suppressing signals of the extreme high and low frequencies, the bandpass filter 3 passes those signals of a specific bandwidth that can be further processed to detect or determine the direction of the sound source. The specific values of the bandwidth depends on the type of target wave source. If the source is a human speaker, the bandwidth may be between 300 Hz and 1500 Hz where typical speech signals have most of their energy concentrated. The bandwidth may also be changed by a calibration process, a trial-and-error process. Instead of using a fixed bandwidth during the operation, initially a certain bandwidth is tried. If too many measurement errors result, the bandwidth is adjusted to decrease the measurement errors so as to arrive at the optimal bandwidth.

5.4 Direction Estimation

For efficiency of computation, the system first finds the approximate direction of the sound source, without the burden of heavy computation, and subsequently calculates the precise direction by using more computation power. The approximate direction is also used to determine the subset of microphones that are relevant to subsequent refinement of the approximate direction. In some configurations, some of the microphones may not have a line of sight to the source, and thus may create phase errors if they participate in further refinement of the approximate direction. Therefore, a subset of microphones are selected that would be relevant to further refinement of the source direction.

5.4.1 Approximate-Direction Finding

FIG. 3 shows the approximate-direction finder 4 in detail. It is based on the idea of specifying the approximate direction of the sound source in terms of a direction perpendicular to a pair of microphones. Let peripheral microphone pairs be the microphones located adjacent to each other around the periphery of the structure holding the microphones, except that a microphone located at the center of the structure, if any, are excluded. For each peripheral microphone pair, "pair direction" is defined as the direction in the horizontal plane, pointing from the center of the pair outward from the structure, perpendicular to the line connecting the peripheral microphone pair.

"Sector direction" is then defined as the pair direction closest to the source direction, selected among possible pair directions. If there are n pairs of peripheral microphones, there would be n candidates for the sector direction.

The sector direction corresponding to the sound source is determined using a zero-delay cross-correlation. For each peripheral microphone pair, a correlation calculator 31 calculates a zero-delay cross-correlation of two signals received from the microphone pair, $X_i(t)$ and $X_j(t)$. It is known to those skilled in the art that such a zero-delay cross-correlation function, $R_{ij}(0)$, over a time period T can be defined by the following formula:

$$R_{ij}(0) = \sum_{t=0}^{T} X_i(t)\, X_j(t)$$

It is noted that a correlation calculator is well-known to those skilled in the art and may be available as an integrated circuit. Otherwise, it is well-known that such a correlation calculator can be built using discrete electronic components such as multipliers, adders, and shift registers.

Among the peripheral microphone pairs, block 32 finds the sector direction by selecting the microphone pair that produces the maximum correlation. Since the signals having the same or similar phase are correlated with each other, the result is to find the pair with the same phase (equi-phase) or having the least phase difference. Since the plane of equi-phase is perpendicular to the propagation direction

of the sound wave, the pair direction of the maximum correlation pair is, then, the sector direction, i.e., the pair direction closest to the source direction.

Once the sector direction is found, block 33 identifies the microphones that participate in further refinement of the approximate direction. "Sector" is defined as the subset of the microphones in the microphone array, which participate in calculating the precise direction of the sound source. For example, where some of the microphones in the array are blocked by a mechanical structure, the signals received by those microphones are not likely to be from direct-travelling waves, and thus such microphones should be excluded from the sector.

In one preferred embodiment, the sector includes the maximum-correlation peripheral microphone pair, another peripheral microphone adjacent to the pair, and a center microphone, if any. Of two peripheral microphones adjacent to the maximum-correlation peripheral microphone pair, the one with a higher zero-delay cross-correlation is selected. The inclusion of the center microphone is optional, but the inclusion helps to improve the accuracy of the source direction, because otherwise three adjacent microphones would be arranged almost in a straight line. There may be other ways of selecting the microphones to be included in the sector, and the information about such selection schemes may be stored in computer memory for an easy retrieval during the operation of the system.

5.4.2 Precise-Direction Finding

The precise-direction finder 5 calculates the precise direction of the sound source using a full cross-correlation. Block 41 first identifies all possible combinations of microphone pairs within the sector. For each microphone pair identified, block 42 calculates a full cross-correlation, $R_{ij}(\tau)$, over a time period T using the follow formula, a well-known formula to those skilled in the art:

$$R_{ij}(\tau) = \sum_{t=0}^{T} X_i(t)\, X_j(t-\tau)$$

As mentioned before, a correlation calculator is well-known to those skilled in the art and may be available as an integrated circuit. Otherwise, it is well-known that such a correlation calculator can be built using discrete electronic components such as

multipliers, adders, and shift registers (Berdugo et al., "On Direction Finding of an Emitting Source From Time Delays," annexed hereto).

$R_{ij}(\tau)$ can be plotted as a cross-correlation curve. For each $R_{ij}(\tau)$, block 43 finds the delay, τ_s corresponding to the peak point of the cross-correlation curve. Note that this peak-correlation delay τ_s lies at a sampling point. In reality, however, the maximum-correlation point may be located between sampling points. Therefore, block 44 calculates such maximum-correlation delay (which may be between sampling points), τ_d, by interpolating the cross-correlation function using a parabolic curve ($y = p\,x^2 + q\,x + r$) as follow:

$$C(k-1) = P\,k^2 + q\,(k-1) + r$$
$$C(k) = P\,k^2 + q\,k + r$$
$$C(k+1) = P\,k^2 + q\,(k+1) + r$$

By solving the above equation for p, q, and r, the maximum point is obtained by obtaining the derivative of the parabolic curve and setting the derivative of the equation to zero. The maximum point τ_d is - (1/2p), and is further expressed as follow:

$$\tau_d = \frac{1}{f_s}\left(k + \left(\frac{C(k-1)-C(k+1)}{2\,(C(k-1) - 2\,C(k) + C(k+1))}\right)\right)$$

where f_s denotes the sampling frequency; k denotes the sampling point corresponding to τ_s; and C(k) is the delay corresponding to sampling point k. The use of the interpolation technique improves the accuracy of the maximum-correlation delay, while eliminating the need for using a very high sampling rate.

Since each maximum-correlation delay calculated for each microphone pair indicates the direction of the sound source measured by individual microphone pairs, the individual maximum-correlation delays are combined to estimate an average direction of the sound source. The estimation process provides a better indication of the source direction than each individual measured directions because it eliminates ambiguity problems inherent to each individual pair and provides a mechanism to verify the relevancy of the individual measurements by possibly eliminating those individual measurements that are far off from the source direction.

Block 45 calculates the precise direction of the sound source in terms of a vector in the Cartesian coordinates, $K = (K_x, K_y, K_z)$, from the vector of

individual measured delays, T_d, by solving the linear equation between K and T_d. The time delay between any two sensors is equal to the projection of the distance vector between them along the K vector divided by the sound velocity. Thus, the T_d vector can be expressed as follows:

$$T_d = - (R K) / c$$

where c is the speed of sound; R denotes the matrix representing the geometry of the microphone array in terms of position differences among the microphones as follows:

$$R = \begin{bmatrix} X_2-X_1, Y_2-Y_1, Z_2-Z_1 \\ \dots \\ X_M-X_1, Y_M-Y_1, Z_M-Z_1 \end{bmatrix}$$

Since the above equation is over-determined in that there are more constraints than the number of variables, the least-square (LS) method is used to obtain the optimal solution. Defining the error as the difference between the measured time delay vector and the evaluated time delay calculated, the error vector ε is given by:

$$\varepsilon = (R K / c) + T_d$$

The solution depends on the covariance matrix Λ of the delay measurements which is defined by

$$\Lambda = E\{T_d T_d^T\} - E\{T_d\}E\{T_d\}^T = COV\{T_d\}$$

where $E\{\}$ denotes the expected value operator, and $\{*\}^T$ denotes the transpose of a matrix. The LS estimated solution, K, is then expressed in the following formula:

$$K = - c (R^T \Lambda^{-1} R)^{-1} R^T \Lambda^{-1} T_d$$
$$= - c B T_d$$

where $\{*\}^{-1}$ denotes the inverse of a matrix. For derivation of the equation, see A. Gelb, Applied Optimal Estimation, the M.I.T. Press, Cambridge, Massachusetts, 1974, e.g., at page 103.

Note that the B matrix depends only on the geometry of the microphone array, and thus can be computed off-line, without burdening the computation requirement during the direction determination.

Block 46 converts K into polar coordinates. FIG. 5 shows the 3-dimensional coordinate system used in the present invention. An azimuth angle, ϕ, is defined as the angle of the source direction in the horizontal plane, measured clockwise from a reference horizonal direction (e.g. x-axis). An elevation angle, Θ, is

defined as the vertical angle of the source direction measured from the vertical axis (z-axis).

Block 46 calculates ϕ and Θ from K_x, K_y, and K_z by converting the Cartesian coordinates to the polar coordinates by solving the nonlinear equation between (K_x, K_y, K_z) and (ϕ, Θ):

$$\begin{bmatrix} K_x \\ K_y \\ K_z \end{bmatrix} = \begin{bmatrix} \sin(\Theta)\cos(\phi) \\ \sin(\Theta)\sin(\phi) \\ \cos(\Theta) \end{bmatrix}$$

In the case of a 3-dimensional microphone array (with the upper microphone), the above equation yields three non-linear equations with two unknowns (ϕ, Θ). The problem is over-determined that there are more equations than the number of variables. The LS solution for (ϕ, Θ) has no close-form solution, but a suboptimal, closed-form, estimation can be found as:

$$\phi = \tan^{-1}(K_y/K_x)$$

$$\Theta = \tan^{-1}(\sqrt{(K_x^2+K_y^2)}/K_z)$$

If a 2-dimensional microphone array were used (without the upper microphone), block 46 calculates ϕ and Θ from K_x and K_y using the following formula:

$$\phi = \tan^{-1}(K_y/K_x)$$

$$\Theta = \cos^{-1}(\sqrt{1-(K_x^2+K_y^2)})$$

Note that the algorithm can function even when the microphones are arranged in a 2-dimensional arrangement and still capable of resolving the azimuth and elevation.

5.5 Measurement Qualification Unit

When the precise-direction finder 5 calculates the precise-direction of the sound source, the result may not reflect the true direction of the sound source due to various noise and measurement errors. The purpose of the measurement qualification unit 6 is to evaluate the soundness or validity of the precise direction using a variety of measurement criteria and invalidate the measurements if the criteria are not satisfied.

FIGS 6a, 6b, 6c, and 6d show different embodiments of the measurement qualification unit using a different measurement criterion. These embodiments may be used individually or in any combination.

FIG. 6a shows a first embodiment of the qualification unit that uses a signal-to-noise ratio (SNR) as a measurement criterion. The SNR is defined as a ratio of a signal power to a noise power. To calculate the SNR, the measured signals are divided into blocks of signals having a predetermined period such as 40 milliseconds. Block 61 calculates the signal power for each signal block by calculating the square-sum of the sampled signals within the block. The noise power can be measured in many ways, but one convenient way of measuring the noise power may be to pick the signal power of the signal block having the minimum signal power and to use it as the noise power. Block 62 selects the signal block having the minimum power over a predetermined interval such as 2 second. Block 63 calculates the SNR as the ratio of the signal power of the current block to that of the noise power. Block 64 invalidates the precise direction if the SNR is below a certain threshold.

Fig. 6b shows a second embodiment of the measurement qualification unit that uses a spread (range of distribution) of individual measured delays as a measurement criterion. The precise source direction calculated by the precise-direction finder represents an average direction among the individual measured directions measured by microphone pairs in the sector. Since delays are directly related to direction angles, the spread of the individual measured delays with respect to the individual estimated delay indicates how widely the individual directions vary with respect to the precise direction. Thus, the spread gives a good indication as to the validity of the measurements. For example, if the individual measured delays are too widely spread, it is likely to indicate some kind of measurement error.

T_e is defined as a vector representing the set of individual estimated delays τ_e corresponding to the precise direction, K. Block 71 calculates T_e from K based on the linear relation between K and T_e.

$$T_e = (-R K) / c$$

where R denotes the position difference matrix representing the geometry of the microphone array as follows

$$[\, X_2-X_1,\ Y_2-Y_1,\ Z_2-Z_1 \,]$$

15

$$R = \begin{bmatrix} \dots & \\ X_M-X_1, & Y_M-Y_1, Z_M-Z_1 \end{bmatrix}$$

and c is the propagation velocity of sound waves.

Block 72 compares the individual measured delays τ_d with the individual estimated delays τ_e and calculates the spread of individual measured delays using the following measure:

$$\Sigma e^2 = \Sigma (\tau_d - \tau_e)^2$$

If this spread exceeds a certain threshold, block 73 invalidates the precise source direction.

Alternatively, the spread can be calculated directly from the individual measured delays using the following:

$$\Sigma e^2 = E * T_d$$

where $E = R (R^T R)^{-1} R^T - I$; and I is the identity matrix.

FIG. 6c shows a third embodiment of the measurement qualification unit that uses the azimuth angle, ϕ, as a measurement criterion. If ϕ deviates significantly from the sector direction (the approximate source direction), it is likely to indicate that the precise direction is false. Therefore, if ϕ is not within a permissible range of angles (e.g. within +/- 60 degrees) of the sector direction, the precise direction is invalidated.

FIG. 6d shows a fourth embodiment of the measurement qualification unit that uses the elevation angle, Θ, as a measurement criterion. If Θ deviates significantly from the horizontal direction (where $\Theta = 90°$), it is likely to indicate the direction of reflected sound waves through the ceiling or the floor rather than that of direct sound waves. Therefore, if Θ is not within a range of allowable angles (e.g. from 30° to 150°), the precise direction is invalidated.

As mentioned before, the above embodiments can be used selectively or combined to produce a single quality figure of measurement, Q, which may be sent to a target system such as a controller for a videoconferencing system. For example, Q may be set to 0 if any of the error conditions above occurs and set to the SNR otherwise.

The direction finding system of the present invention can be used in combination with a directional microphone system, which may include an adaptive

filter. Such adaptive filter is not limited to a particular kind of adaptive filter. For example, one can practice the present invention in combination with the invention disclosed in applicant's commonly assigned and co-pending U.S. patent application Serial No. 08/672,899, filed June 27, 1996, entitled 'System and Method for Adaptive Interference Cancelling,' by inventor Joseph Marash and its corresponding PCT application WO 97/50186, published December 31, 1997. Both applications are incorporated by reference herein in their entirety.

Specifically, the adaptive filter may include weight constraining means for truncating updated filter weight values to predetermined threshold values when each of the updated filter weight value exceeds the corresponding threshold value. The adaptive filter may further include inhibiting means for estimating the power of the main channel and the power of the reference channels and for generating an inhibit signal to the weight updating means based on normalized power difference between the main channel and the reference channels.

The weight constraining means may include a frequency-selective weight-control unit, which includes a Fast Fourier Transform (FFT) unit for receiving adaptive filter weights and performing the FFT of the filer weights to obtain frequency representation values, a set of frequency bins for storing the frequency representation values divided into a set of frequency bands, a set of truncating units for comparing the frequency representation values with a threshold assigned to each bin and for truncating the values if they exceed the threshold, a set of storage cells for temporarily storing the truncated values, and an Inverse Fast Fourier Transform (IFFT) unit for converting them back to the adaptive filter weights.

The adaptive filter in the directional microphone that may be used in combination with the present invention may also employ dual-processing interference cancelling system where adaptive filter processing is used for a subset of a frequency range and fixed filter processing is used for another subset of the frequency range. For example, one can practice the present invention in combination with the invention disclosed in applicant's commonly assigned and co-pending U.S. patent application Serial No. 08/840,159, filed April 14, 1997, entitled 'Dual-Processing Interference Cancelling System and Method,' by inventor Joseph Marash, corresponding continuation-in-part application, Ser. No. 09/055,709, filed April 7, 1998, and

corresponding PCT Application Ser. No. PCT/IL98/00179, filed April 14, 1998. All three applications are incorporated by reference herein in their entirety.

It is noted that the adaptive filter processing portion of the dual processing may also employ the adaptive filter processing disclosed in applicant's commonly assigned and co-pending U.S. patent application Serial No. 08/672,899, filed June 27, 1996, entitled 'System and Method for Adaptive Interference Cancelling,' by inventor Joseph Marash and its corresponding PCT application WO 97/50186, published December 31, 1997.

5.6 Software Implementation

The present invention described herein may be implemented using a commercially available digital signal processor (DSP) such as Analog Device's 2100 Series or any other general purpose microprocessors. For more information on Analog Device 2100 Series, see Analog Device, ADSP-2100 Family User's Manual, 3rd Ed., 1995.

FIGS. 8A-8D show a flow chart depicting the operation of a program in accordance with a preferred embodiment of the present invention. The program uses measurement flags to indicate various error conditions.

When the program starts (step 100), it resets the system (step 101) by resetting system variables including various measurement flags used for indicating error conditions. The program then reads into registers microphone inputs sampled at the sampling frequency of 64 KHz (step 102), which is oversampling over the Nyquist frequency. As mentioned in Section 5.2, oversampling allows anti-aliasing filters to be realized with a much more gentle cut-off characteristic of a filter. Upon reading every 5 samples (step 103), the program performs a low-pass filter operation and a decimation by taking one sample out of every 5 samples for each microphone (step 104). The decimated samples are stored in the registers (step 105).

The program performs a bandpass filter operation on the decimated samples so that the output contains frequencies ranging from 1.5 to 2.5 KHz (step 106). The output is stored in input memory (step 107). The program repeats the above procedure until 512 new samples are obtained (step 108).

If the 512 news samples are reached, the program takes each pair of adjacent microphone pairs and multiples the received signals and add them to obtain

zero-delay cross-correlation (step 200), and the results are stored (step 206). The calculation of zero-delay cross-correlation is repeated for all adjacent microphone pairs, not involving the center microphone (step 201).

The microphone pair having the highest zero-delay cross-correlation is selected (step 202) and the value is stored as the signal power (step 207), which will be used later. Of those two microphones adjacent to the selected pair, the program calculates the zero-correlation (step 203) and the microphone having the higher correlation is selected (step 204). The program determines the sector by including the selected microphone pair, the neighboring microphone selected, and the center microphone, if there is one.

The program calculates the average power of the 512 samples taken from the center microphone (step 300). The lowest average energy during the latest 2 seconds is set to be the noise power (steps 301-305).

The program calculates the full cross-correlation of signals received by each microphone pair in the sector (step 306). The program finds the peak cross-correlation delay, τ_s, where the correlation is maximum (step 307).

τ_s lies on a sampling point, but the actual maximum-correlation delay, τ_d, may occur between two sampling points. If τ_s is either the maximum or minimum possible delay (step 308), τ_d is set to τ_s (step 309). Otherwise, the program finds the actual maximum-correlation delays using the parabolic interpolation formula described in Section 5.4.1 (steps 310-312). The above steps are repeated for all the microphone pairs in the sector (step 313).

The program uses the B matrix mentioned in Section 5.4.2 to obtain the direction vector $K = [K_x, K_y, K_z]$ from the set of time delays (step 400).

The program then calculates the azimuth angle, ϕ, and the elevation angle, Θ, corresponding to the direction vector obtained (step 401).

The program calculates the SNR as the ratio of the signal power and the noise power (step 402). If the SNR exceeds a threshold (step 403), the program raises the SNR Flag (step 404).

The program then evaluates the elevation angle, Θ. If Θ is not within a permissible range of angles (e.g. from 30° to 150°) (step 405), the Elevation Flag is raised (step 406).

The program calculates corresponding delays from the precise direction (step 407). The program calculates a delay spread as the sum of squares of the difference between the individual measured delays and the individual estimated delays (step 408). If the delay spread exceeds a certain threshold (step 409), the Delay Spread Flag is raised (step 410).

The program calculates the quality figure of measurement, Q, as a combination of all or part of the measurement criteria above (step 411). For example, Q may be set to 0 if any of the measurement flags was raised and set to the SNR otherwise.

The program transfers ϕ, Θ, and Q to a target system, such as an automatic camera tracking system used in a video conferencing application (step 412). The program resets the measurement flags (step 413) and goes back to the beginning of the program (step 414).

While the invention has been described with reference to several preferred embodiments, it is not intended to be limited to those embodiments. It will be appreciated by those of ordinary skill in the art that many modifications can be made to the structure and form of the described embodiments without departing from the spirit and scope of the invention, which is defined and limited only in the following claims. For example, the present invention can be used to locate a direction of a source transmitting electromagnetic waves.

ANNEX

"On direction finding of an emitting source from time delays"

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Abstract

This paper presents a statistically and computationally efficient algorithm for direction finding of a single far field source using a multi-sensor array. The algorithm extracts the azimuth and elevation angles directly from the estimated time delays between the array elements. Hence, it is referred to herein as the Time Delay Direction Finding (TDDF) algorithm. An asymptotic performance analysis, using a small error assumption, is conducted. For any 1-D and 2-D array configurations, it is shown that the TDDF algorithm achieves the Cramer Rao Lower Bound (CRLB) for the azimuth and elevation estimates provided that the noise is Gaussian and spatially uncorrected and that the time delay estimator achieves the CRLB as well. Moreover, with the suggested algorithm no constrains on the array geometry are required. For the general 3-D case the algorithm does not achieve the CRLB for a general array. However it is shown that for array geometries which obey certain constraints the CRLB is achieved as well.

The TDDF algorithm offers several advantages over the beamforming approach. First, it is more efficient in terms of computational load. Secondly, the azimuth estimator does not require the a-priory knowledge of the wave propagation velocity. Thirdly, the TDDF algorithm is suitable for applications where the arrival time is the only measured input, in contrast to the beamformer, which is not applicable in this case.

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1. Introduction

In various applications of array signal processing such as radar, sonar and seismology, there is a great interest in detection and localization of wideband sources. The problem of estimating the direction of arrival (DOA) of wideband sources using a sensor array has been studied extensively in the literature [2-14]. A common approach[2-7] to this problem, for a single source scenario, is to use the time delay estimation between two sensors to determine the DOA. Many techniques for estimating the travel time delay between *two* receiving sensors have been investigated , see *e.g.* [2-7]. For the *single* source and a *multi-sensor* case, Hahn and Tretter[8] introduced the Maximum Likelihood (ML) delay-vector estimator. ML DOA estimators for the multi sensors and multi sources case have also being studied extensively [12-14]

It is well known [11] that the ML DOA estimator, for the single source case with a spatially uncorrelated noise, can be realized as a focused beamformer. In this paper an alternative approach is proposed, in which the DOA is extracted directly from the estimated time delays between the array elements (referred to as the time delay vector). This approach is an extension to the multi-sensor case, of the work in[10,11], where the DOA is extracted from the time delay between *two* sensors for the far field case.

The suggested Time Delay Direction Finding (TDDF) algorithm utilizes the linear relationship between the time delay vector and the DOA vector in Cartesian coordinates. This linear relationship allows a closed form estimation of the DOA vector. The transformation to polar coordinates i.e. azimuth and elevation is straightforward for 1-D and 2-D array geometries . For arbitrarily chosen 3-D array

configurations, the best estimator requires a simple non-linear least squares minimization. Alternatively, a closed form suboptimal solution for the 3-D configuration is also suggested. Finally it is shown that the TDDF azimuth and elevation estimator achieves the CRLB provided that the time delay estimators achieves the CRLB as well

2. Methods

2.1 The Time Delay Direction Finding (TDDF) Algorithm:

Consider an array of M identical omni-directional sensors with a known arbitrary geometry measuring the wavefield generated by a single farfield wideband source in the presence of an additive noise. Let $\bar{r}_i$ denote the location of the i-th sensor, where $\bar{r}_i = [x_i, y_i, z_i]$ for the 3-D array, $\bar{r}_i = [x_i, y_i]$ for the 2-D case, and $\bar{r}_i = [x_i]$ for the 1-D case, and let ϕ and θ denote the azimuth and elevation angles of the radiating source, respectively (see Fig. 1).

Let us now define the differential delay vector,

$$\bar{\tau} = \left[\tau_{12}, \tau_{13}, \ \ldots \ , \tau_{1,M}\right]^T; \qquad \tau_{1j} \equiv \tau_j - \tau_1, \tag{1}$$

where the first sensor serves as a reference. The signal DOA vector for the far field case is given by:

$$\bar{k} = \begin{bmatrix} k_x \\ k_y \\ k_z \end{bmatrix} = \begin{bmatrix} \sin(\theta)\cos(\phi) \\ \sin(\theta)\sin(\phi) \\ \cos(\theta) \end{bmatrix}. \tag{2}$$

The time delay between any two sensors is equal to the projection of the distance vector between them along the $\vec{k}$ vector divided by the sound velocity. Consequently, the delay vector can be expressed as follows:

$$\vec{\tau} = -\frac{R\vec{k}}{c}; \qquad R \equiv \begin{bmatrix} \vec{r}_2 - \vec{r}_1 \\ \vdots \\ \vec{r}_M - \vec{r}_1 \end{bmatrix} \qquad (3)$$

where c is the wave velocity and the matrix R is composed of the distance vectors between all the sensors and the reference sensor.

The objective is to estimate $\vec{k}$ from the measured time delay vector $\vec{\tau}$. Studying Eq.(3), it is evident that the problem is overdetermined. Thus, it is suggested to apply the least squares (LS) method to obtain the estimation. Defining the error as the difference between the measured time difference vector and the evaluated time vector (calculated from the assumed $\vec{k}$ vector), the error vector is given by:

$$\vec{\varepsilon} = \left(\frac{R\vec{k}}{c} + \hat{\vec{\tau}} \right) \qquad (4)$$

In the general case, the measurement errors of the time delay vector need not be uncorrelated. Hence, the solution depends on the covariance matrix Λ_τ of the delays measurements which is defined by,

$$\Lambda_\tau = E\left\{ \hat{\vec{\tau}} \hat{\vec{\tau}}^T \right\} - E\left\{ \hat{\vec{\tau}} \right\} E\left\{ \hat{\vec{\tau}} \right\}^T = COV\left\{ \hat{\vec{\tau}} \right\} \qquad , \qquad (5)$$

where E{} denotes the expected value operator. The problem is "over determined" for M>3. The LS solution for $\hat{\vec{k}}$, the DOA vector, in this case is given by[3]:

$$\hat{\vec{k}} = \underset{\vec{k}}{\text{ArgMin}} \left\{ \left(\frac{R\vec{k}}{c} + \hat{\vec{\tau}} \right)^T \Lambda_\tau^{-1} \left(\frac{R\vec{k}}{c} + \hat{\vec{\tau}} \right) \right\} \qquad (6)$$

$$= -c\left(\mathbf{R}^T\Lambda_\tau^{-1}\mathbf{R}\right)^{-1}\mathbf{R}^T\Lambda_\tau^{-1}\hat{\bar{\tau}} \equiv -c\mathbf{B}\hat{\bar{\tau}} \quad ,$$

Thus, estimating the DOA vector becomes a simple multiplication between the measured time delay vector $\bar{\tau}$ and a data independent matrix **B** . The matrix **B** depends on the array geometry (through **R**) and the time delay covariance matrix which under the assumption of spatially uncorrelated noise is known a-priory up to a multiplicative factor which cancels out in this equation. Consequently, it can be calculated off-line.

In order to express the DOA vector in terms of azimuth and elevation, one has to write the vector $\hat{\bar{k}}$ in a polar coordinate representation. For a 1-D array configuration only $\hat{k}_x$ can be estimated. Hence, assuming horizontal elevation, the azimuth angle is given by:

$$\hat{\phi} = \cos^{-1}(\hat{k}_x). \tag{7}$$

With a 2-D array, both the azimuth and elevation angles can be calculated by:

$$\hat{\phi} = \tan^{-1}(\hat{k}_y / \hat{k}_x), \tag{8a}$$

$$\hat{\theta} = \cos^{-1}(\hat{k}_z) = \cos^{-1}\left(\left(1-(\hat{k}_x^2+\hat{k}_y^2)\right)^{1/2}\right). \tag{8b}$$

For the case of a 3-D array, Eq.(2) yields three non-linear equations with two unknowns (ϕ,θ). Again the problem is over determined. Thus, the azimuth and elevation angles (ϕ,θ) can be evaluated as the nonlinear least square estimator solving Eq.(2)

$$(\hat{\phi},\hat{\theta}) = \underset{\hat{\phi},\hat{\theta}}{\text{ArgMin}} \left\{\left(\hat{\bar{k}}-\bar{k}(\hat{\phi},\hat{\theta})\right)^T\Lambda_k^{-1}\left(\hat{\bar{k}}-\bar{k}(\hat{\phi},\hat{\theta})\right)\right\}, \tag{9}$$

where Λ_k is the covariance matrix of vector $\hat{\mathbf{k}}$, which is given in Appendix A (Eq.(14)).

An alternative simplified close-form suboptimal estimate is proposed by:

$$\hat{\phi} = \tan^{-1}(\hat{k}_y / \hat{k}_x), \tag{10a}$$

$$\hat{\theta} = \tan^{-1}\left(\left(\hat{k}_x^2 + \hat{k}_y^2\right)^{1/2} / \hat{k}_z\right). \tag{10b}$$

In appendix A. the performance of the TDDF algorithm is analyzed, and it is shown that it is asymptotically efficient. Furthermore, it is shown that under certain geometrical constrains for the sensors arrangement even the closed form 3-D solution achieves the CRLB .

Importantly, it should be noted, that the azimuth estimates given above for the 2-D and 3-D array configurations, are independent of the wave velocity c, (stems from the fact that the solutions are given in terms of the ratio between $\hat{k}_y$ and $\hat{k}_x$, and both are a linear function of c). Therefore, errors in the assumed sound speed will not induce errors in the azimuth angle.

2.2 Performance analysis

In appendix A. the covariance matrix for $(\hat{\phi},\hat{\theta})$ is calculated. The performance of the TDDF estimator is compared to the theoretical CRLB. It is shown that for the 1-D and 2-D cases the estimator is asymptotically efficient since it achieves the bound. For the 3-D case the closed form estimator given in Eq.(10) is not always efficient. Howeverwe we derived constrains on the array geometry in which the CRLB is also achieved.

3. Results

In this section, the performance of the TDDF algorithm is demonstrated via numerical simulations and by experimental results.

3.1 Numerical Simulations

Simulations were conducted for 2-D and 3-D arrays. The 2-D array was comprised of randomly located 7 sensors, as shown in Fig. 2(a). In the first set of simulations the source was positioned at a fixed location with an azimuth angle of 60^0 and an elevation of 30^0. The SNR was scanned in the range of -10dB to +10dB, the integration time was $50ms$, and the frequency bandwidth was 500-1500Hz. The noisy estimates of the time delay vectors were generated as Gaussian random vectors with a covariance matrix Λ_τ given in equation Eq.(12). The propagation velocity was taken to be 340m/sec.

Five hundred Monte Carlo runs were performed for each SNR value. The azimuth angle was calculated with the TDDF algorithm and the corresponding errors were computed. The standard deviation of the localization errors was then estimated. For comparison, the CRLB was also calculated as explained in appendix A. The results are depicted in Fig.3. For clarity of presentation, only 50 azimuth estimation errors are plotted (as small dots) at each SNR level. The standard deviation of the TDDF estimator are depicted as circles and the corresponding CRLB is depicted as a solid line for the entire range. It can be seen that the standard deviations of the TDDF estimator are effectively located on the CRLB line (in accordance with the mathematical derivation given in appendix A).

In a second set of simulations, the same 2-D array was used. The SNR was held fixed at -6dB. The elevation angle was set to 50^0, and the azimuth angle was scanned in the range of $0^0 - 360^0$. The corresponding CRLB line was calculated and the standard deviation of the TDDF estimator was evaluated for each angle. Figures 4(a) and 4(b) display the errors of the TDDF algorithm for the azimuth and elevation, respectively. From these curves it is observed that the TDDF estimator achieves the CRLB, both in azimuth and elevation, for a 2-D array with an arbitrary geometry.

In the third set of simulations the 3-D array shown in Fig. 2(b) was used. This array has 6 sensors equally distributed on a circle with a radius of $0.1m$ and the seventh sensor is located $0.1m$ above the center of the array. The SNR was again held fixed at -6dB. The azimuth was set to 20^0, and the elevation angle was scanned in the range $0^0 - 180^0$. The following quantities were calculated this time: the CRLB, the standard deviation of the TDDF estimator, and the theoretical standard deviation calculated from Eq.(25). Figures 5(a) and 5(b) plot these quantities as a function of the elevation angle.

The 3-D array used here obeys the condition given in Eq.(27). Consequently, the CRLB is achieved for the azimuth TDDF estimates (Fig.5(a)). For the elevation angle, however, the obtained estimate errors are larger than the CRLB. This observation is consistent with the fact that the array geometry does not comply with condition given in Eq.(29). Nevertheless, the degradation is moderate for this array configuration. This implies that the closed form estimation given by equation set .(10) is sufficiently accurate for practical purposes. Finally it can be observed that the estimated errors match the theoretical standard deviations predicted by Eq.(25). It should also be noted that bias was also estimated in all the above simulation and was

found negligible (two orders of magnitude smaller than the variance contribution to the total error).

In practical applications it is usually not easy to obtain the optimal estimate for the time delay vector τ. In the last numerical example we demonstrate the performance of the TDDF algorithm when using a suboptimal estimator for the time delay vector as presented in appendix C. The time delay vector was estimated via a cross-correlation between the reference sensor (#1) and the other sensors. The performance of the TDDF algorithm is compared to that of a beamformer. This simulation uses the 2-D array consisting of 3 microphones which is shown in Fig 2c. The source direction was set at an azimuth of 30° and an elevation angle of 90°. The SNR was scanned in the range of -10dB to +10dB. Here, the simulation generates the time record of the sensor data assuming spatially uncorrected noise. Both the signal and the noise were random Gaussian variables with a bandwidth of 100-3000 Hz. In order to perform the beam steering required in the beamformer the data were interpolated by a factor of 10. The standard deviation error of both estimators was estimated by 100 Monte Carlo runs. The standard deviation of the TDDF estimate are depicted by asterisks in figure 6 while the standard deviation of the Beamformer is denoted by circles . As can be seen for most of the studied SNR range the performance of the TDDF is the same as that of the beamformer. However, the threshold point for the TDDF appears at SNR=-3db which is higher than the threshold observed for the beamformer (-6db). This result is not surprising since the TDDF is not an ML estimator as the beamformer. Potentially there are two factors that can cause the performance of the TDDF to collapse. The first is the time delays vector estimation process, and the second is the nonlinear operation for estimating γ. In all

the cases we have tested, the time delay estimate was the first one to diverge. Practically it was observed that the cross-correlation function have generated spurious peaks at low SNR, and this is probably the main cause for the performance diverges at low SNR.

3.2 Experimental Results

A 3-D array consisting of 7 microphones (AudioTechnica MT350B) arranged in the same configuration depicted in Fig2(b), (Radius=0.1m', height=0.1m'), was used. Two experiments were conducted. In the first experiment the array was located in an unechoic chamber (internal dimension of 1.7 x1.7x1.7 m'). In the second experiment the array was placed in an ordinary room. The sound source was a recorded male voice (Richard Burton) reading a 20-second long sentence. The signal was played via a loudspeaker located 1.5 m' from the array. The outputs of the array were recorded using an 8-channel tape recorder (Sony-pc208A). The time delay between the sensors and the central microphone was estimated by filtering the data by a band pass filter 500-1500 Hz, and performing a cross-correlation process. The integration time was 40 ms, yielding about 500 independent measurements to estimate the system performance. After completion of each set of measurements, the array was rotated by 30^0 and the procedure was repeated.

The azimuth angles corresponding to each set of measurements was estimated using the TDDF algorithm. The standard deviation of the errors for the TDDF estimates was then evaluated. The results are outlined in Fig. 7 . The data from the unechoic chamber is denoted by 'o' and the data from the regular room is presented by '*'. As can be observed the average TDDF error for the unechoic chamber

experiment was about 1.5^0. The second experiment was held in the regular room, and the average error was about 5^0. This degradation is attributed to the room reverberations and the background noise. We have measured the reverberation time in both rooms. In the unechoic chamber the reverberation time was about 10-ms, while in the regular room the reverberation time was about 250-ms. Thus, we believe that the room reverberation was the major cause for the degradation in the accuracy of the direction estimates.

4. Discussion

This paper presents and analyzes the Time Delay Direction Finding (TDDF) algorithm for a single emitting source using a multi-sensor array. The algorithm extracts the azimuth and elevation angles directly from the estimated time delays between the array elements. The algorithm offers computational simplicity as it utilizes the linear relationship between the time delay vector and the DOA vector in Cartesian coordinates. This linear relationship allows a closed form estimation of the DOA vector.

An asymptotic performance analysis of the TDDF algorithm, using a small error assumption is performed. For the 1-D and 2-D array configurations it is shown that the TDDF algorithm achieves the Cramer Rao Lower Bound (CRLB) provided that the time delay vector estimator achieves the CRLB as well. This was proven mathematically in appendix A and was demonstrated by numerical simulations. For a 3-D array configuration a suboptimal closed form estimator is presented (Eq.(10)). Nevertheless, it is shown that when using array geometries that obey certain constrains

the closed form solution also achieves the CRLB. If the array obeys the condition given by Eq.(27), then the azimuth estimation is statistically efficient. Furthermore, if the array also obeys the constrains that are given in Eq.(29), the estimator is efficient for the elevation angle as well.

Numerical and experimental results were given to demonstrate the performance of the TDDF algorithm. The experimental results with a 7 microphone array have shown that in an unechoic chamber the average TDDF azimuth error was about 1.5 degrees, while in a regular room the average error was about 5 degrees. These results indicate that the TDDF can serve as a practical tool for passive localization of a single radiating source.

The proposed TDDF algorithm offers several advantages over the popular beamforming approach[11] . First, the TDDF algorithm is considerably more efficient in terms of computational load. It calculates the azimuth and the elevation angle directly from the estimated time delays, and does not involve a two dimensional search over the array manifold as the beamformer.

Secondly, for the 2-D and 3-D array configurations, the TDDF algorithm does not require the a-priory knowledge of the propagation velocity to estimate the azimuth, see Eq.(8a) and Eq.(10a). respectively. This property of the TDDF is very useful in acoustic applications where uncertainty in the propagation velocity occurs due to wind and temperature variations [17]. This is contrary to the beamformer , which uses the wave propagation velocity as input. In principle the beamforming process could scan the velocity as an additional unknown parameter, however this would substantially increase the computational load as an additional parameter would have to be scanned.

The third advantage of the TDDF method, arises in applications where the signal is a very short transient and the arrival time of the pulse is directly measured by the system hardware. Since only the time delays are available in this case the beamforming is not applicable. For TDDF algorithm on the other hand this information is sufficient.

Finally, in certain acoustic and geophysical applications, loss of spatial coherence of the signal received at the sensors may occur if the distance between the sensors is large[1, 17], thus precluding the use of the beamforming approach. In such cases, nevertheless, time delay between sensors can still be estimated via incoherent processing means, such as time of arrival difference, and the TDDF algorithm is still applicable.

The TDDF has one major disadvantage. It is limited to a single source scenario. The beamformer algorithm on the other hand can localize more then one source, provided that the angular separation between the sensors is more then the beam-width.

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Appendix A

Performance Analysis of the TDDF Algorithm

In order to evaluate the performance of the TDDF algorithm, analytic expressions for the accuracy of the azimuth and elevation estimations for the 1-D, 2-D and 3-D cases are first derived. These expressions are then compared to the expression of the CRLB as derived by Neilsen.[16] and cited in Appendix B.

For uniformity and simplicity of notations let us define $\gamma = \phi$ for the 1-D case, and $\gamma = (\phi, \theta)^T$ for the 2-D and 3-D cases. Under the assumption of small errors the covariance matrix of γ can be expressed as:

$$\Lambda_\gamma = \overline{\nabla}_x \gamma \cdot \Lambda_x \cdot \overline{\nabla}_x^T \gamma = \overline{\nabla}_x \gamma \cdot \overline{\nabla}_\tau k \cdot \Lambda_\tau \cdot \overline{\nabla}_\tau^T k \cdot \overline{\nabla}_x^T \gamma \qquad (11)$$

where Λ_x is the covariance matrix of the vector x, and $\overline{\nabla}_x y$ is the gradient (Jacobian) of $\bar{y}$ with respect to $\bar{x}$. Again for a small error assumption it can be verified that the DOA estimates are asymptotically unbiased, thus the covariance matrix represents the total error of the estimator.

Clearly, the performance of the TDDF algorithm depends on the covariance matrix of the time delay vector. To demonstrate the performance of the TDDF algorithm we shall assume that the time delay vector estimator achieves the CRLB. An efficient algorithm for estimation of the time delay vector assuming that both the signal and the noise are zero mean uncorrelated Gaussian processes, and the noise is spatially uncorrelated, has been proposed and studied by Hahn and Tretter[8]. Their work presents an estimator for the time delay vector which achieves the CRLB, and does not requires the beamformer process, Their scheme is based on estimating M(M-1) individual time delays via a pre-filtered correlators. The vector τ is obtained by a linear combination of the individual time delays. The covariance matrix for the

time delay vector for this estimator, assuming that the SNR is the same for all the sensors, is given by:

$$\Lambda_\tau = \mathrm{CRLB}_\tau = \frac{\sigma_\tau^2}{2}\left[\mathbf{I}_{M-1} + \mathbf{1}_{M-1}\mathbf{1}_{M-1}^T\right] \tag{12}$$

Where: $\mathbf{I}_M$ is the MxM Identity matrix; $\overline{1}_M$ is an M-dimensional vector of ones,

$\sigma_\tau^2 = \left(\sum_{l=1}^{L_{max}} (\omega_0 l)^2 \rho(l) \frac{M\rho(l)}{1+M\rho(l)} \right)^{-1}$, and $\rho(l) \equiv S(l)/N(l)$ denotes the SNR at the frequency $(\omega_0 l)$.

In the following it shall be assumed that Λ_τ is given by Eq.(12) i.e. efficient estimate of $\bar{\tau}$,

In appendix B we derive Λ_τ for suboptimal time delay estimator via only (M-1) correlators, using one sensor as a reference sensor i.e. an efficient estimate is only obtained for the separate pairwise delays. It is shown that for sufficiently high SNR this estimator also achieves the CRLB.

First, the covariance matrix of the direction vector $\bar{k}$ is calculated,

From Eq.(6)

$$\overline{\nabla}_\tau^T k = c\mathbf{B} \tag{13}$$

and therefore

$$\Lambda_k = E\left\{\overline{\mathbf{k}\mathbf{k}}^T\right\} = \mathbf{B}\Lambda_\tau \mathbf{B}^T c^2 \tag{14}$$

Using the definition of **B** in Eq.(6) and applying some algebraic simplifications yields,

$$\Lambda_k = \left(\mathbf{R}\Lambda_\tau^{-1}\mathbf{R}^T\right)^{-1} c^2 . \tag{15}$$

Applying the matrix inversion lemma to Eq.(12) it can be written that

$$\Lambda_\tau^{-1} = \frac{2}{\sigma_\tau^2}\left[\mathbf{I}_{M-1} - \frac{\bar{1}_{M-1}\bar{1}_{M-1}^T}{M}\right]. \tag{16}$$

From the definitions of **R** in Eq.(3) it follows that

$$\mathbf{R} = [-1_{M-1} \quad \mathbf{I}_{M-1}]\mathbf{P} \tag{17}$$

Where **P** is the sensor position matrix defined by:

$$\mathbf{P} = [\bar{x}, \bar{y}, \bar{z}] = \begin{bmatrix} x_1 & y_1 & z_1 \\ : & : & : \\ x_M & y_M & z_M \end{bmatrix} \tag{18}$$

Substituting Eq.(16) and Eq.(17) into Eq.(15) yields after some algebraic manipulations:

$$\Lambda_k = \left(\mathbf{P}^T\mathbf{P} - \frac{\left(\mathbf{P}^T\bar{1}_M\right)\left(\mathbf{P}^T\bar{1}_M\right)^T}{M}\right)^{-1}\frac{\sigma_\tau^2}{2}c^2. \tag{19}$$

Assuming without any loss of generality that the coordinate origin is in the center of gravity i.e. $\mathbf{P}^T\bar{1}_M = \bar{0}$ we finally get the simple expression for the DOA vector covariance matrix,

$$\Lambda_k = \left(\mathbf{P}^T\mathbf{P}\right)^{-1}\frac{\sigma_\tau^2}{2}c^2 \tag{20}$$

In the following, the expressions for the accuracy of the azimuth and elevation TDDF estimations are derived and compared to the CRLB which is cited in appendix B.

A.1 A linear (1-D) array configuration

For the 1-D case $\gamma = \phi$, and $\tilde{k} = k_x = \cos(\phi)$. Thus

$\overline{\nabla}_k \gamma = -1/\sin(\phi)$; and $\Lambda_k = \frac{1}{\sum x_i^2} \frac{\sigma_r^2}{2} c^2$ inserting into Eq.(11) gives

$$\sigma_\phi^2 = \frac{1}{\sin^2(\phi) \sum x_i^2} \frac{\sigma_r^2}{2} c^2 \tag{21}$$

The CRLB for the 1-D case is given by: $CRLB(\phi) = 1/J_{\phi\phi}$ (see appendix B). Substituting $y_i=z_i=0$ in this expressions yields:

$$CRLB(\phi) = \frac{1}{\sin^2(\phi) \sum x_i^2} \frac{\sigma_r^2}{2} c^2$$

As can be observed this expression is identical to the right hand side of Eq.(21), indicating that the TDDF estimate achieves the CRLB in this case.

A.2 A planar (2-D) array configuration

From Eq.(2) and Eq.(8) the Jacobian $\overline{\nabla}_k \gamma$ is given by :

$$\overline{\nabla}_k \gamma = \frac{1}{-\sin(\theta)\cos(\theta)} \begin{bmatrix} -\sin(\phi)\cos(\theta) & \cos(\phi)\cos(\theta) \\ -\cos(\phi)\sin(\theta) & -\sin(\phi)\sin(\theta) \end{bmatrix}. \tag{22}$$

Inserting Eq.(22) and Eq.(20) into Eq.(11) yields,

$$\sigma_\phi^2 = \frac{1}{\sin^2(\theta)} \frac{\cos^2(\phi)\sum x_i^2 + 2\cos(\phi)\sin(\phi)\sum x_i y_i + \sin^2(\phi)\sum y_i^2}{\sum x_i^2 \sum y_i^2 - \left(\sum x_i y_i\right)^2} \frac{\sigma_\tau^2}{2} c^2. \qquad (23a)$$

and

$$\sigma_\theta^2 = \frac{1}{\cos^2(\theta)} \frac{\sin^2(\phi)\sum x_i^2 - 2\cos(\phi)\sin(\phi)\sum x_i y_i + \cos^2(\phi)\sum y_i^2}{\sum x_i^2 \sum y_i^2 - \left(\sum x_i y_i\right)^2} \frac{\sigma_\tau^2}{2} c^2 . \qquad (23b)$$

Applying a lengthy but straightforward evaluating of the expressions for the CRLB Eq.(A1) for both the azimuth and the elevation angles for the 2-D arrays, i.e. $z_i=0$; shows that they are identical to Eq.(23). Thus, it is concluded that the TDDF algorithm is a statistically efficient estimator for 2-D array, which reaches the CRLB. It is important to note that no constraints on the array geometry were applied.

A.3 A spatial (3-D) array configuration

The estimate of $\gamma = (\phi, \theta)$ for the 3-D array case involves a nonlinear LS minimization Eq.(9). An alternate close form close-form suboptimal estimator was suggested in Eq.(10). Here we calculate the performance of the sub-optimal estimator and derive the conditions on the array geometry that guarantee statistical efficiency (achieves the CRLB).

From Eq.(2) and Eq.(10) the Jacobian ∇_γ is given by:

$$\nabla_\gamma = \begin{bmatrix} -\frac{\sin(\phi)}{\sin(\theta)} & \frac{\cos(\phi)}{\sin(\theta)} & 0 \\ \cos(\theta)\cos(\phi) & \cos(\theta)\sin(\phi) & -\sin(\theta) \end{bmatrix} \qquad (24)$$

Using Eq.(11) and Eq.(15)

$$\sigma_\phi^2 = \frac{\sigma_\tau^2 c^2}{2\sin^2(\theta)} \left[\sin(\phi) \quad \cos(\phi) \quad 0\right] \left(\mathbf{P}^T\mathbf{P}\right)^{-1} \left[\sin(\phi) \quad \cos(\phi) \quad 0\right]^T, \qquad (25a)$$

and

$$\sigma_\theta^2 = \frac{\sigma_\tau^2 c^2}{2}\left[\cos(\theta)\cos(\phi) \quad \cos(\theta)\sin(\phi)) \quad -\sin(\theta)\right]\left(\mathbf{P}^T\mathbf{P}\right)^{-1} \cdot \left[\cos(\theta)\cos(\phi) \quad \cos(\theta)\sin(\phi)) \quad -\sin(\theta)\right]^T \tag{25b}$$

In general this estimator is not efficient. However if the array obeys the following geometrical conditions.

$$\begin{aligned} &\sum x_i y_i = \sum x_i z_i = \sum y_i z_i = 0 \\ &\sum x_i^2 = \sum y_i^2 \end{aligned} \tag{27}$$

then the variance of the azimuth angle estimation is given by

$$\sigma_\phi^2 = \frac{1}{\sin^2(\theta)} \frac{1}{\sum x_i^2} \frac{\sigma_\tau^2}{2} c^2 \tag{28}$$

Evaluating the CRLB for the azimuth estimate under the same condition yields an identical expression. Thus, under the conditions outlined in Eq.(27), the TDDF algorithm is also an efficient estimator for the azimuth angle. When studying the conditions for uncoupled estimates of azimuth and elevation angles, Nielsen[16] has also reached the same conditions given in Eq.(27), and gave a few examples of 3-D arrays obeying these constraints.

If we further constrain the array geometry and require a fully balanced array configuration, which obeys the following condition,

$$\sum x_i^2 = \sum z_i^2 \tag{29}$$

in addition to the conditions outlined in Eq.(27), it can be shown that for the elevation estimate,

$$\sigma_\theta^2 = \frac{1}{\sum x_i^2} \frac{\sigma_\tau^2}{2} c^2 = CRLB_\theta \tag{30}$$

Thus, under the above conditions the TDDF algorithm achieves the CRLB for both the azimuth and the elevation angles, and is therefore an asymptotically efficient estimator.

Appendix B- CRLB for the azimuth and elevation angles

Nilsen[16] derived analytic expressions for the Cramer Rao Lower Bound for the estimation errors of the azimuth angle ϕ and the elevation angle θ, using 3-D arrays.

$$CRLB(\theta) = J_{\phi\phi}/(J_{\phi\phi}J_{\theta\theta} - J_{\phi\theta}^2)$$
$$CRLB(\phi) = J_{\theta\theta}/(J_{\phi\phi}J_{\theta\theta} - J_{\phi\theta}^2) \qquad (31)$$

where

$$J_{\phi\phi} = G\sin^2(\theta)\sum_{l=1}^{M}\left[x_l\sin(\phi) - y_l\cos(\phi)\right]^2$$
$$J_{\theta\theta} = G\sum_{l=1}^{M}\left[x_l\cos(\phi)\cos(\theta) + y_l\sin(\phi)\cos(\theta) - z_l\sin(\theta)\right]^2$$
$$J_{\phi\theta} = G\sin(\theta)\sum_{l=1}^{M}\left[x_l\sin(\phi) - y_l\cos(\theta)\right]\left[x_l\cos(\phi)\cos(\theta) + y_l\sin(\phi)\cos(\theta) - z_l\sin(\theta)\right]$$
$$G = \sum_{l=1}^{L_{max}}(\omega_0 l)\frac{M\rho^2(l)}{1 + M\rho(l)}$$

In these expressions the coordinates origin is in the center of gravity of the array, i.e.

$$\sum x_l = \sum y_l = \sum z_l = 0.$$

and the coordinate system is given in Fig. 1

Appendix C Suboptimal estimation for the time delays vector

In this appendix we consider a suboptimal estimation of the time delays vector which estimates only M-1 time delays between the first sensor relative to all the other sensors in the array. Each time delay estimation is the based on the data of these two sensors only, and ignores the fact it is part of an M sensors array. Efficient estimate for the time delay between two sensors can be obtains by maximizing the generalized cross-correlation[2].

Based on the derivation in[8] the covariance matrix of this estimator is given by:

$$\Lambda_\tau = \frac{1}{\left\{\sum_{k=1}^{K_{max}} 2\frac{(\omega_0 l)^2\rho^2}{1+2\rho}\right\}^2}\sum_{k=1}^{K_{max}} 2\frac{(\omega_0 l)^2\rho^2}{(1+2\rho)^2}\begin{bmatrix} 1+2\rho & . & \rho & . \\ . & 1+2\rho & . & . \\ \rho & . & . & . \\ . & . & . & 1+2\rho \end{bmatrix} \qquad (32)$$

In general this estimator does not achieves the CRLB, however for high SNR case $2\rho >> 1$ it can be seen that

$$\Lambda_\tau \approx \frac{1}{2\cdot\sum_{k=1}^{K_{max}}(\omega_0 l)^2\rho}\left[I_{M-1} + 1_{M-1}1_{M-1}^T\right]$$

Evaluating the $CRLB_\tau$ as given in equation Eq.(12) for the high SNR case yields the same expression. i.e. for a good SNR case the suboptimal time delays estimator achieves the $CRLB_\tau$.

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Figure Captions

Figure 1: Schematic representation of the model and the coordinate system used here.

Figure 2: (a).The 2-D array geometry consisting of 7 randomly located microphones, which was used in the first two numerical simulations. (b) The 3-D array geometry, which was used in the third numerical simulation and the experimental measurements. (c) The 2-D array consisting of 3 microphones which was used in the last numerical simulation.

Figure 3: Azimuth estimation errors of the simulated 2-D array, as a function of the SNR. The source is positioned at an azimuth of 60° and an elevation angle of 30°. The solid line is the CRLB. The dots depict the magnitude of the errors of the first 25 individual runs (out of the 500 used). The circles depict the standard deviation of the TDDF estimator

Figure 4: The errors of the TDDF algorithm for the azimuth (a) and elevation (b) as a function of the azimuth, for the 2-D array shown in Fig. 2(a). The SNR is -6 dB, the elevation angle is 50^0. The solid line is the CRLB, the circles depict the standard deviation of the TDDF estimator

Figure 5: The errors of the TDDF algorithm for the azimuth (a) and elevation (b) as a function of the elevation angle, for the 3-D array shown in Fig.2(b). The SNR is -6 dB, the azimuth is 20^0. The solid line is the CRLB. The circles depict the standard deviation of the TDDF estimator, and the analytic variance expressions (Eq.23) are depicted as '+'.

Figure 6: Azimuth estimation errors of the 3 microphone 2-D array used in the last simulation, as a function of the SNR. The source is positioned at an azimuth of 30°. The standard deviation of the TDDF estimate are depicted by '*' and the standard deviation of the Beamformer is plotted by 'o'.

Figure 7 : Experimentally measured azimuth errors of the TDDF algorithm as a function of the azimuth using the 3-D array. The source was a speech signal played from a loud speaker 1.5m' away from the array. The circles 'o' denotes the results measured in an unechoic chamber, and the '*' indicates the results measured in an ordinary room.

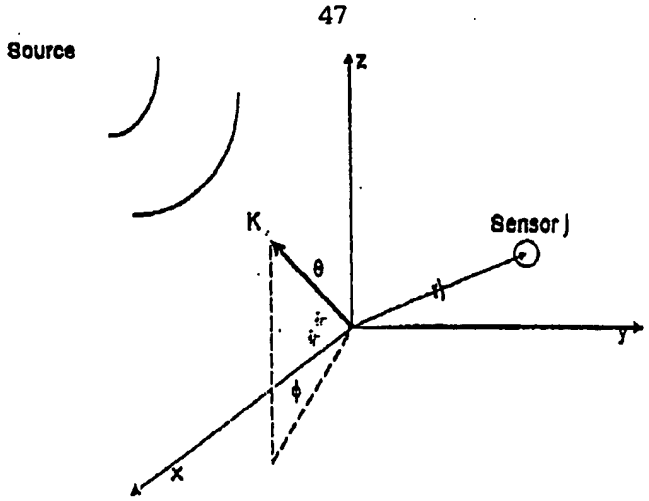

Figure 1
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

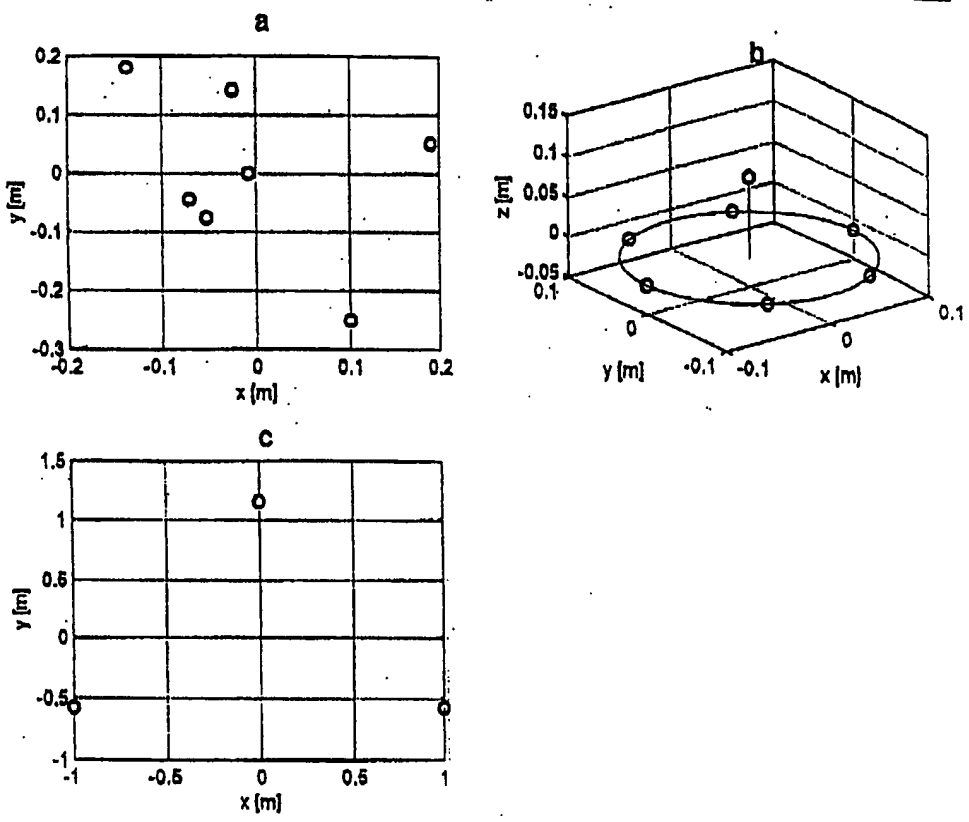

Figure 2
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

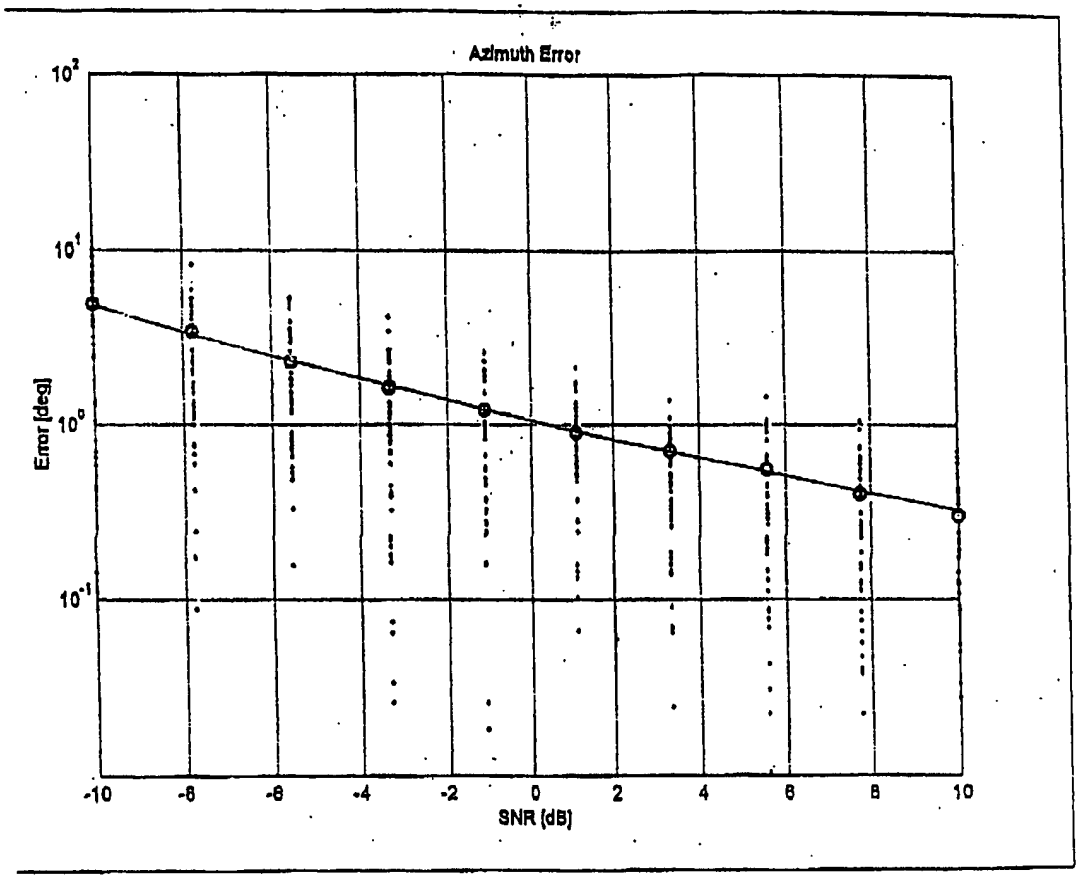

Figure 3
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

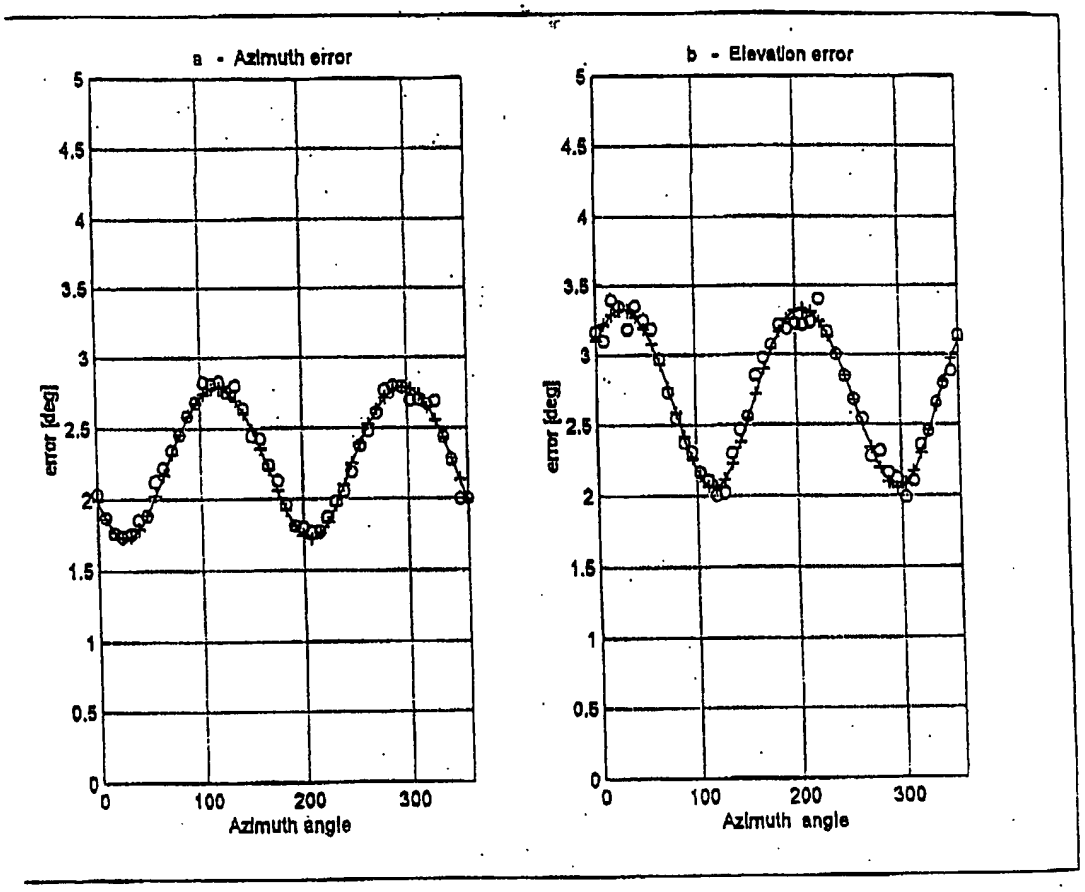

Figure 4
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

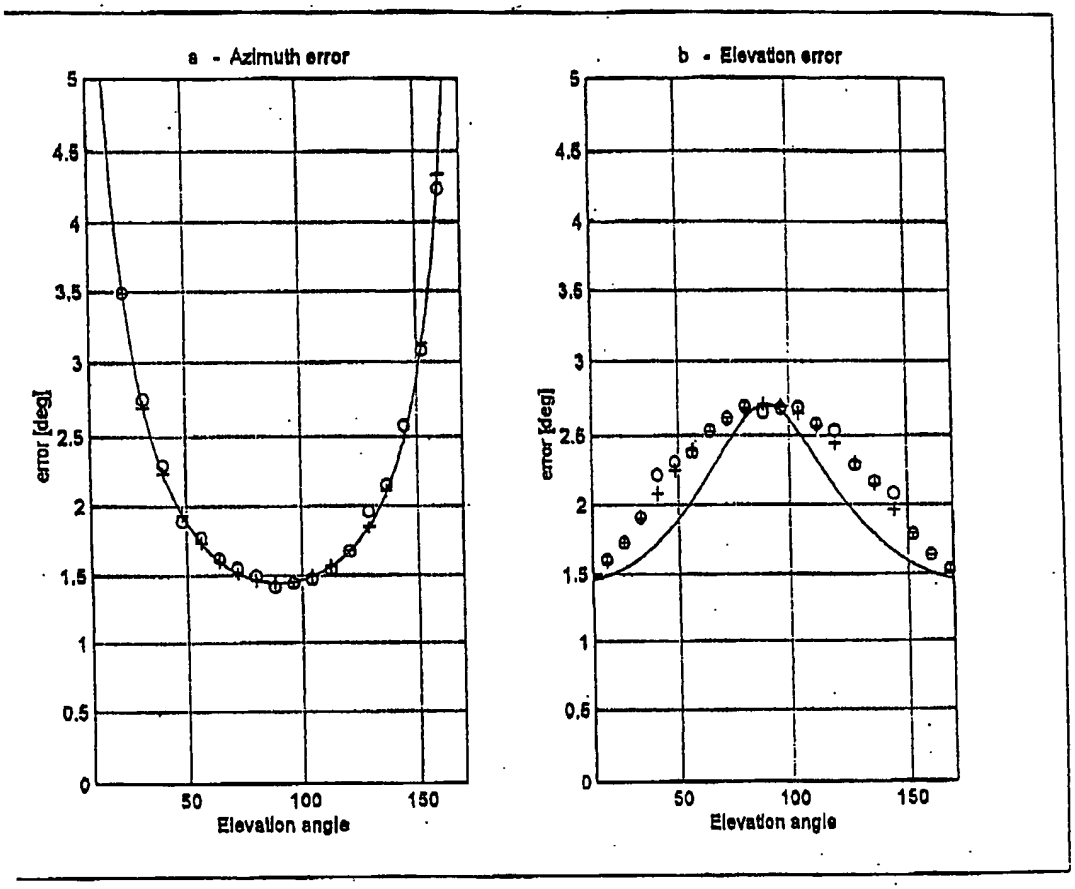

Figure 5
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

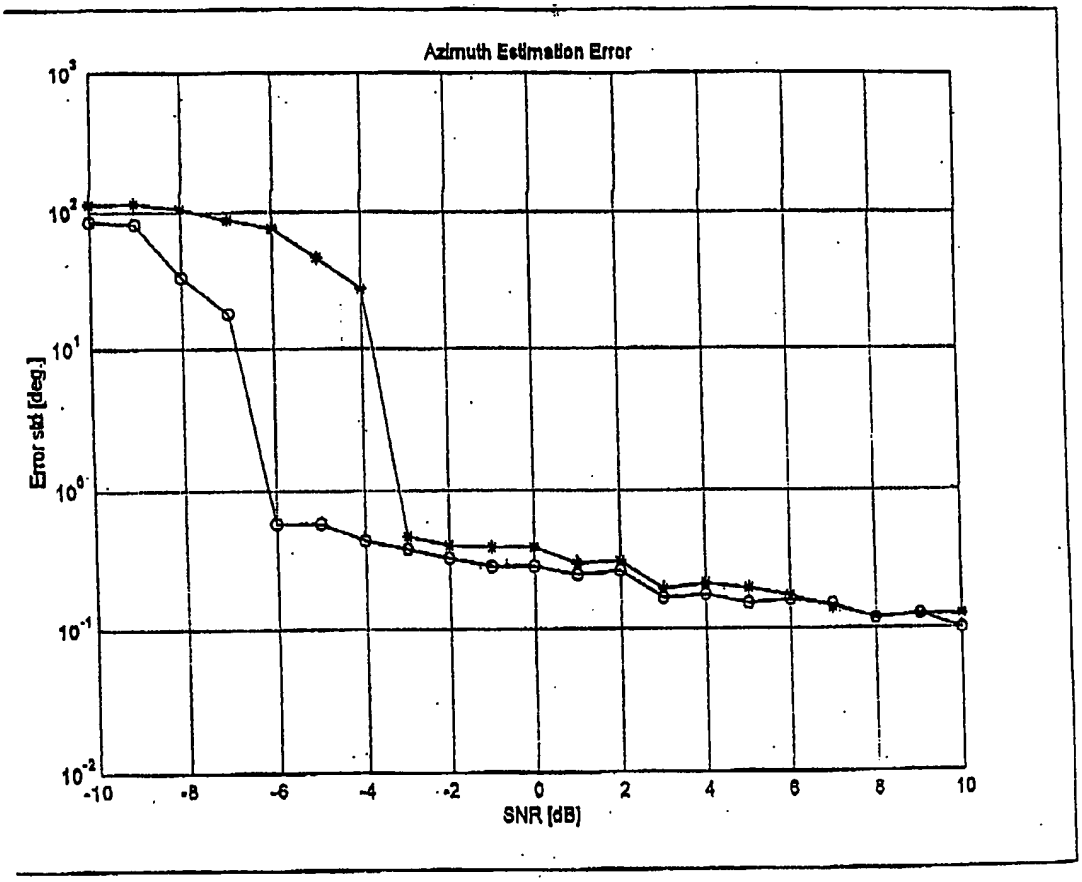

Figure 6
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

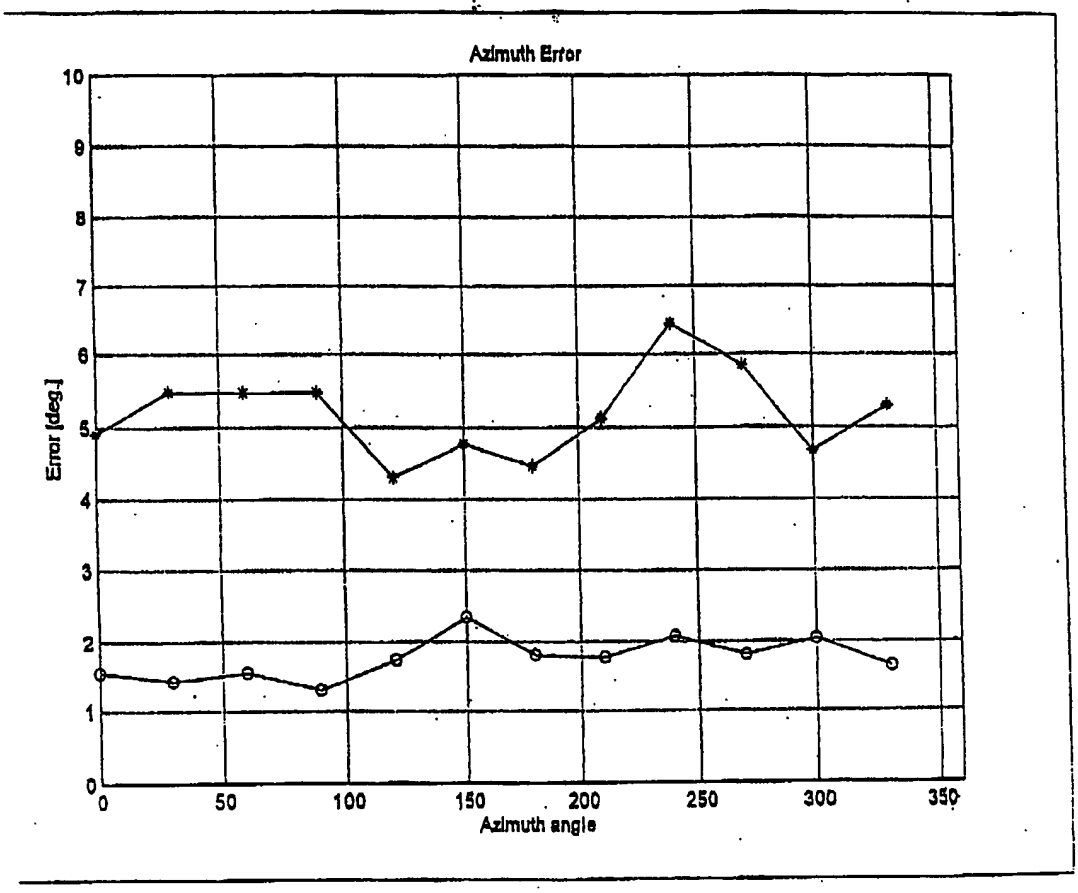

Figure 7
Berdugo et. Al. JASA
"On direction finding of an emitting source from time delays"

What is claimed is:

1. A system for finding the direction of a wave source, comprising:

an array of sensors arranged in a predetermined geometry, each sensor for sensing waves from the wave source and generating signals representing the waves;

an approximate-direction finder, connected to receive the signals representing the waves, for processing the signals to find the approximate direction of the wave source in terms of the positions of a selected subset of sensors;

a precise-direction finder, connected to receive information from the approximate-direction finder, for finding the precise direction of the wave source by further processing the signals representing the waves based on the approximate direction; and

a measurement qualification unit, connected to the precise-direction finder, for evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

2. The system of claim 1, wherein the wave source is a sound source, and the sensors are microphones.

3. The system of claim 1, further comprising:

a bandpass filter, connected to receive signals from the array of sensors, for filtering the signals representing the waves to generate filtered signals containing frequencies of a specific bandwidth.

4. The system of claim 1, wherein the measurement qualification unit comprises:

at least means for calculating a signal-to-noise ratio (SNR); and

means for invalidating the precise direction if the SNR is below a threshold.

5. A system for finding the direction of a wave source, comprising:

an array of sensors arranged in a predetermined geometry, each sensor for sensing waves from the wave source and generating analog signals representing the waves;

an analog-to-digital converter, connected to the array of sensors, for converting the analog signals to digital signals;

a bandpass filter for filtering the digital signals to generate filtered signals containing frequencies of a specific bandwidth;

an approximate-direction finder, connected to receive the filtered signals, for processing the filtered signals to find the approximate direction of the wave source in terms of a sensor pair selected among the sensors;

a precise-direction finder, connected to receive information from the approximate-direction finder, for finding the precise direction of the wave source by further processing the signals representing the waves based on the approximate direction; and

a measurement qualification unit, connected to the precise-direction finder, for evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

6. The system of claim 5, wherein the array of sensors in a predetermined geometry includes sensors arranged in a circular arrangement.

7. The system of claim 5, wherein the wave source is a sound source and the sensors are microphones.

8. The system of claim 5, wherein the measurement qualification unit comprises:

at least means for calculating a signal-to-noise ratio (SNR); and

means for invalidating the precise direction if the SNR is below a threshold.

9. The system of claim 8, wherein the means for calculating a signal-to-noise ratio comprises:

means for calculating a signal power for a current signal block of the measured signals;

means for calculating a noise power by finding the minimum signal power over signal blocks within a predetermined time period; and

means for calculating the signal-to-noise ratio by calculating the ratio of the signal power of a signal block to the noise power.

10. The system of claim 8, wherein the precise direction is calculated in terms of an azimuth angle and an elevation angle.

11. The system of claim 10, wherein the measurement qualification unit further comprises:

means for invalidating the precise direction of the wave source if the elevation angle is not within allowable values.

12. The system of claim 10, wherein the approximate-direction finder comprises:

means for calculating a zero-delay cross-correlation for pairs of sensors adjacent to each other; and

means for identifying a sector direction by selecting the sensor pair having the highest zero-delay cross-correlation.

13. The system of claim 12, wherein the measurement qualification unit comprises:

means for comparing the estimated azimuth angle with the sector direction; and

means for invalidating the precise direction of the wave source if the difference between the azimuth angle and the sector direction is not within allowable values.

14. The system of claim 8, wherein the precise-direction finder comprises:

means for identifying all sensor pairs within the sector;

means for calculating individual measured delays by calculating a full cross-correlation for every pair of sensors within the sector; and

means for finding the precise direction by finding a least-square solution of the individual measured delays.

15. The system of claim 14, wherein the measurement qualification unit comprises:

means for generating individual estimated delays from the precise direction;

means for calculating a delay spread by calculating differences between the individual measured delays and the individual estimated delays; and

means for invalidating the precise direction of the wave source if the delay spread exceeds a threshold.

16. A system for finding the direction of a sound source, comprising:

an array of microphones arranged in a predetermined geometry, each microphone for sensing sound waves from the sound source and generating analog signals representing the sound waves;

an analog-to-digital converter, connected to the array of microphones, for converting the analog signals to digital signals;

a bandpass filter for filtering the digital signals to generate filtered signals containing frequencies of a specific bandwidth;

an approximate-direction finder, connected to receive the filtered signals, for processing the filtered signals to find the approximate direction of the sound source in terms of a sensor pair selected among the sensors;

a precise-direction finder, connected to receive information from the approximate-direction finder, for finding the precise direction of the sound source by further processing the signals representing the sound waves based on the approximate direction; and

a measurement qualification unit for evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

17. The system of claim 16, wherein the array of microphones in a predetermined geometry includes microphones arranged in a circular arrangement.

18. The system of claim 16, wherein the microphones are omni microphones.

19. The system of claim 16, wherein the microphones are cardioid microphones.

20. The system of claim 16, wherein the microphones are dipole microphones.

21. The system of claim 16, wherein the specific bandwidth is determined by the frequency range of sound waves of interest to direction determination.

22. The system of claim 16, wherein the specific bandwidth is determined by a calibration process of dynamically adjusting the bandwidth to arrive at the optimal bandwidth.

23. The system of claim 16, wherein the measurement qualification unit comprises:

at least means for calculating a signal-to-noise ratio (SNR); and

means for invalidating the precise direction if the SNR is below a threshold.

24. The system of claim 23, wherein the means for calculating a signal-to-noise ratio comprises:

means for calculating a signal power for a current signal block of the measured signals;

means for calculating a noise power by finding the minimum signal power over signal blocks within a predetermined time period; and

means for calculating the signal-to-noise ratio by calculating the ratio of the signal power of a signal block to the noise power.

25. The system of claim 23, wherein the precise direction is calculated in terms of an azimuth angle and an elevation angle.

26. The system of claim 25, wherein the measurement qualification unit further comprises:

means for invalidating the precise direction of the wave source if the elevation angle is not within allowable values.

27. The system of claim 26, wherein the allowable values are between 30 to 150 degrees.

28. The system of claim 25, wherein the approximate-direction finder comprises:

means for calculating a zero-delay cross-correlation for pairs of sensors adjacent to each other; and

means for identifying a sector direction by selecting the sensor pair having the highest zero-delay cross-correlation.

29. The system of claim 28, wherein the measurement qualification unit comprises:

means for comparing the estimated azimuth angle with the sector direction; and

means for invalidating the precise direction of the wave source if the difference between the azimuth angle and the sector direction is not within allowable values.

30. The system of claim 29, wherein the allowable values are between -60 and 60 degrees.

31. The system of claim 23, wherein the precise-direction finder comprises:

means for identifying all sensor pairs within the sector;

means for calculating individual measured delays by calculating a full cross-correlation for every pair of sensors within the sector; and

means for finding the precise direction by finding a least-square solution of the individual measured delays.

32. The system of claim 31, wherein the measurement qualification unit comprises:

means for generating individual estimated delays from the precise direction;

means for calculating a delay spread by calculating differences between the individual measured delays and the individual estimated delays; and

means for invalidating the precise direction of the wave source if the delay spread exceeds a threshold.

33. A computer for processing digital signals representing a sound source, sampled from an array of microphones sensing sound waves from the sound source and for finding the direction of the sound source, comprising:

a memory containing:

a program for finding the approximate direction of the sound source by processing the digital signals to find the approximate direction of the sound source in terms of a sensor pair selected among the sensors;

a program for finding the precise direction of the sound source by further processing the digital signals based on the approximate direction; and

a program for measurement qualification by evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

34. The computer of claim 33, wherein the program for finding the approximate direction of the sound source comprises:

a program for calculating a zero-delay cross-correlation for pairs of sensors adjacent to each other; and

a program for identifying a sector direction by selecting the sensor pair having the highest zero-delay cross-correlation.

35. The computer of claim 33, wherein the program for finding the precise-direction of the sound source comprises:

a program for identifying all sensor pairs within the sector;

a program for calculating individual measured delays by calculating a full cross-correlation for every pair of sensors within the sector; and

a program for finding the precise direction by finding a least-square solution of the individual measured delays.

36. The computer of claim 33, wherein the program for measurement qualification comprises:

at least a program for calculating a signal-to-noise ratio (SNR); and

a program for invalidating the precise direction if the SNR is below a threshold.

37. A system for controlling the direction of a microphone to the direction of a particular sound source, comprising:

an array of microphones arranged in a predetermined geometry, each microphone for sensing sound from the sound source and generating signals representing the sound;

an approximate-direction finder, receiving the signals representing the sound, for calculating the approximate direction of the sound source in terms of a pair of microphones selected among the microphones;

a precise-direction finder, connected to the approximate-direction finder, for finding the precise direction of the sound source based on the approximate direction found;

a measurement qualification unit for generating a quality figure of measurement by evaluating validity of the precise direction using a measurement criterion; and

a controller for controlling the movement of the microphone using the precise direction of the sound source and the quality figure.

38. The system of claim 37, wherein the microphone system is a directional microphone system.

39. The system of claim 38, wherein the directional microphone system comprises an adaptive filter for suppressing interference.

40. The system of claim 39, wherein the directional microphone system further comprises an array of microphones.

41. The system of claim 40, wherein the adaptive filter has a weight-constraining unit where frequency representations of filter weights are constrained to a threshold to suppress directional interferences.

42. The system of claim 38, wherein the directional microphone further comprises a dual-processing interference cancelling system comprising:

an adaptive-processing filter for processing a first portion of a frequency band; and

a fixed-processing filter for processing a second portion of the frequency band.

43. The system of claim 42, wherein the adaptive filter has a weight-constraining unit where frequency representations of filter weights are constrained to a threshold to suppress directional interferences.

44. A system for controlling the direction of a camera to the direction of a particular sound source, comprising:

an array of microphones arranged in a predetermined geometry, each microphone for sensing sound from the sound source and generating signals representing the sound;

an approximate-direction finder, receiving signals representing the sound, for finding the approximate direction of the sound source in terms of a pair of microphones selected among the microphones;

a precise-direction finder, connected to the approximate-direction finder, for finding the precise direction of the sound source based on the approximate direction found;

a measurement qualification unit for generating a quality figure of measurement by evaluating validity of the precise direction using a measurement criterion; and

a controller for controlling the movement of camera using the precise direction of the sound source and the quality figure.

45. A method of finding the direction of a wave source, comprising:

generating signals representing waves from the wave source;

processing the signals to find the approximate direction of the wave source in terms of a sensor pair selected among the sensors;

finding the precise direction of the wave source by further processing the signals representing the waves based on the approximate direction; and

qualifying measurements by evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

46. The method of claim 45, wherein the wave source is a sound source, and the sensors are microphones.

47. The method of claim 45, further comprising the step of:

filtering the signals representing the waves to generate filtered signals containing frequencies of a specific bandwidth.

48. The method of claim 45, wherein the step of qualifying measurements comprises:

at least the step of calculating a signal-to-noise ratio (SNR); and

the step of invalidating the precise direction if the SNR is below a threshold.

49. A method of finding the direction of a sound source, comprising the steps of:

generating analog signals representing waves from the sound source using an array of sensors arranged in a predetermined geometry, each sensor for sensing the waves;

converting the analog signals to digital signals;

filtering the digital signals to generate filtered signals containing frequencies of a specific bandwidth;

processing the signals to find the approximate direction of the wave source in terms of a sensor pair selected among the sensors;

finding the precise direction of the wave source by further processing the signals representing the waves based on the approximate direction; and

qualifying measurements by evaluating the validity of the precise direction of the wave source using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

50. The method of claim 49, wherein the step of qualifying measurements comprises:

at least the step of calculating a signal-to-noise ratio (SNR); and

the step of invalidating the precise direction if the SNR is below a threshold.

51. The method of claim 50, wherein the step of calculating a signal-to-noise (SNR) ratio comprises:

calculating a signal power for a current signal block of the measured signals;

calculating a noise power by finding the signal block having the minimum signal power over signal blocks within a predetermined time period; and

calculating the signal-to-noise ratio (SNR) by calculating the ratio of the signal power to the noise power.

52. The method of claim 50, wherein the precise direction is calculated in terms of an azimuth angle and an elevation angle.

53. The method of claim 52, wherein the step of qualifying measurement further comprises:

invalidating the precise direction of the wave source if the elevation angle is not within allowable values.

54. The method of claim 53, wherein the allowable values are between 30 to 150 degrees.

55. The method of claim 52, wherein the step of finding the approximate direction comprises:

calculating a zero-delay cross-correlation for pairs of sensors adjacent to each other; and

identifying the sector direction by selecting the sensor pair having the highest zero-delay cross-correlation.

56. The method of claim 55, wherein the step of qualifying measurement comprises:

comparing the estimated azimuth angle with the sector direction; and

invalidating the precise direction if the difference between the azimuth angle and the sector direction is not within allowable values.

57. The method of claim 56, wherein the allowable values are between -60 and 60 degrees.

58. The method of claim 49, wherein the step of finding the precise direction comprises:

identifying all sensor pairs within the sector;

calculating individual measured delays by calculating a full cross-correlation for every pair of sensors within the sector; and

finding the precise direction of the sound source by finding a least-square solution of the individual measured delays.

59. The method of claim 58, wherein the step of qualifying measurement comprises:

generating individual estimated delays from the precise direction;

calculating a delay spread by finding differences between the individual measured delays and the individual estimated delays; and

invalidating the precise direction if the delay spread exceeds a threshold.

60. The method of claim 49, wherein the microphones are omni microphones.

61. The method of claim 49, wherein the microphones are cardioid microphones.

62. The method of claim 49, wherein the microphones are dipole microphones.

63. The method of claim 49, wherein the specific bandwidth is determined by the frequency range of sound waves of interest to direction determination.

64. The method of claim 49, wherein the specific bandwidth is determined by a calibration process of dynamically adjusting the bandwidth to arrive at the optimal bandwidth.

65. A method for processing digital signals representing a sound source, sampled from an array of microphones sensing sound waves from the sound source and for finding the direction of the sound source in a computer having a memory, comprising the steps of:

finding the approximate direction of the sound source by processing the digital signals to find the approximate direction of the sound source in terms of a sensor pair selected among the sensors;

finding the precise direction of the sound source by further processing the digital signals based on the approximate direction; and

qualifying measurements by evaluating validity of the precise direction using a measurement criterion and invalidating the precise direction if the measurement criterion is not met.

66. The method of 65, wherein the step of finding the approximate direction of the sound source comprises the steps of:

calculating a zero-delay cross-correlation for pairs of sensors adjacent to each other; and

identifying a sector direction by selecting the sensor pair having the highest zero-delay cross-correlation.

67. The method of claim 65, wherein the step of finding the precise-direction of the sound source comprises the steps of:

identifying all sensor pairs within the sector;

calculating individual measured delays by calculating a full cross-correlation for every pair of sensors within the sector; and

finding the precise direction by finding a least-square solution of the individual measured delays.

68. The method of claim 65, wherein the step of qualifying measurements comprises:

at least the step of calculating a signal-to-noise ratio (SNR); and

the step of invalidating the precise direction if the SNR is below a threshold.

69. A method of controlling the direction of a microphone system to the direction of a particular sound source, comprising:

generating signals representing sound waves from the sound source using an array of microphones arranged in a predetermined geometry, each microphone for sensing the sound waves;

processing the signals to find the approximate direction of the wave source in terms of a sensor pair selected among the sensors;

finding the precise direction of the wave source by further processing the signals representing the waves based on the approximate direction found; and

generating a quality figure of measurement by evaluating validity of the precise direction using a measurement criterion; and

controlling the movement of the microphone system using the precise direction of the sound source and the quality figure.

70. The method of claim 69, wherein the microphone system is a directional microphone system.

71. The method of claim 70, wherein the directional microphone system comprises an adaptive filter for suppressing interference.

72. A method of controlling the direction of a camera to the direction of a particular sound source, comprising:

generating signals representing sound waves from the sound source using an array of microphones arranged in a predetermined geometry, each microphone for sensing the sound waves;

processing the signals representing sound waves to find the approximate direction of the wave source in terms of a sensor pair selected among the sensors;

finding the precise direction of the wave source by further processing the signals representing waves based on the approximate direction; and

generating a quality figure of measurement by evaluating validity of the precise direction using a measurement criterion; and

controlling the movement of the camera using the precise direction of the sound source and the quality figure.

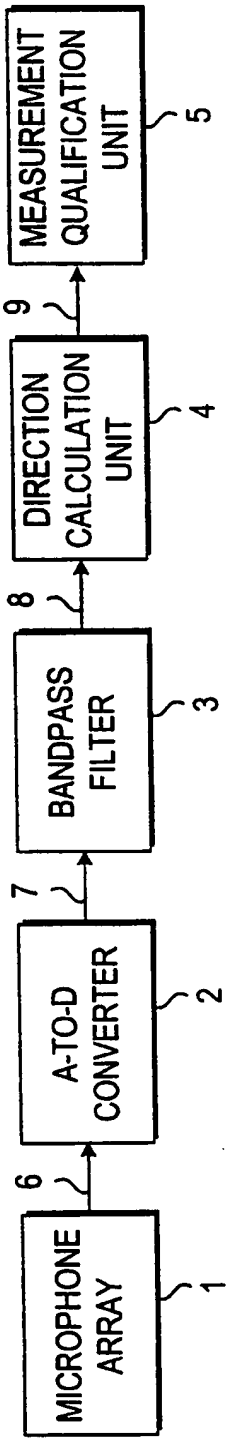
MICROPHONE ARRAY
1
6
A-TO-D CONVERTER
2
7
BANDPASS FILTER
3
8
DIRECTION CALCULATION UNIT
4
9
MEASUREMENT QUALIFICATION UNIT
5

FIG. 1

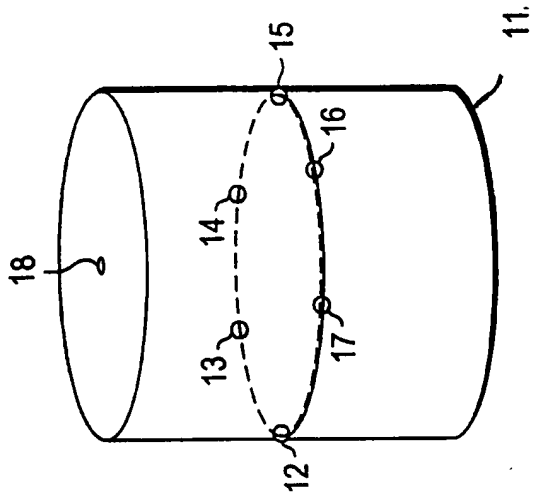

FIG. 2

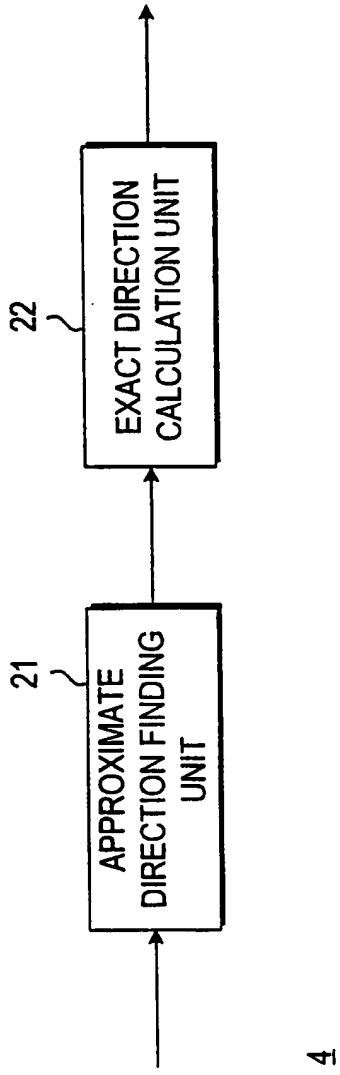
APPROXIMATE DIRECTION FINDING UNIT
21
EXACT DIRECTION CALCULATION UNIT
22
4

FIG. 3

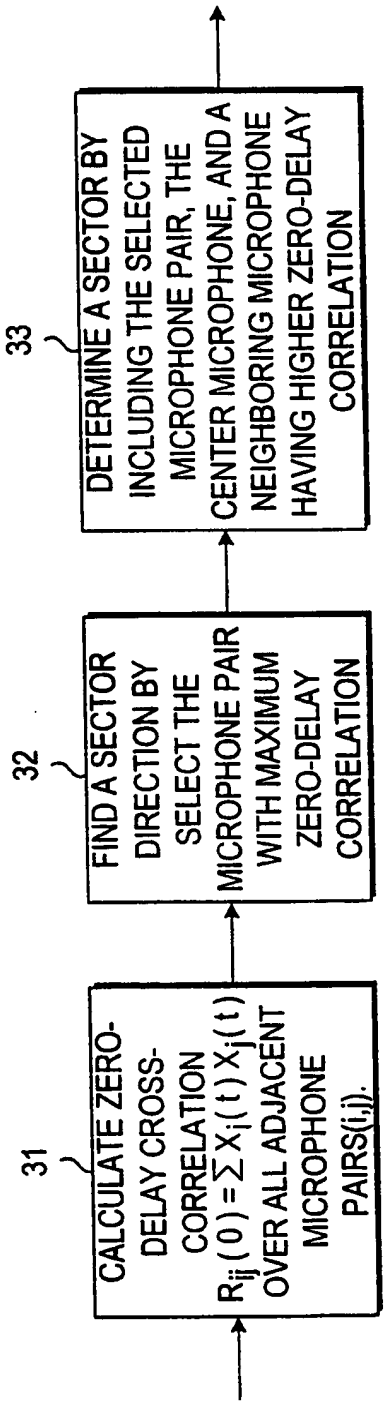
31
CALCULATE ZERO-DELAY CROSS-CORRELATION $R_{ij}(0) = \sum X_i(t) X_j(t)$ OVER ALL ADJACENT MICROPHONE PAIRS(i,j).
32
FIND A SECTOR DIRECTION BY SELECT THE MICROPHONE PAIR WITH MAXIMUM ZERO-DELAY CORRELATION
33
DETERMINE A SECTOR BY INCLUDING THE SELECTED MICROPHONE PAIR, THE CENTER MICROPHONE, AND A NEIGHBORING MICROPHONE HAVING HIGHER ZERO-DELAY CORRELATION
21

FIG. 4

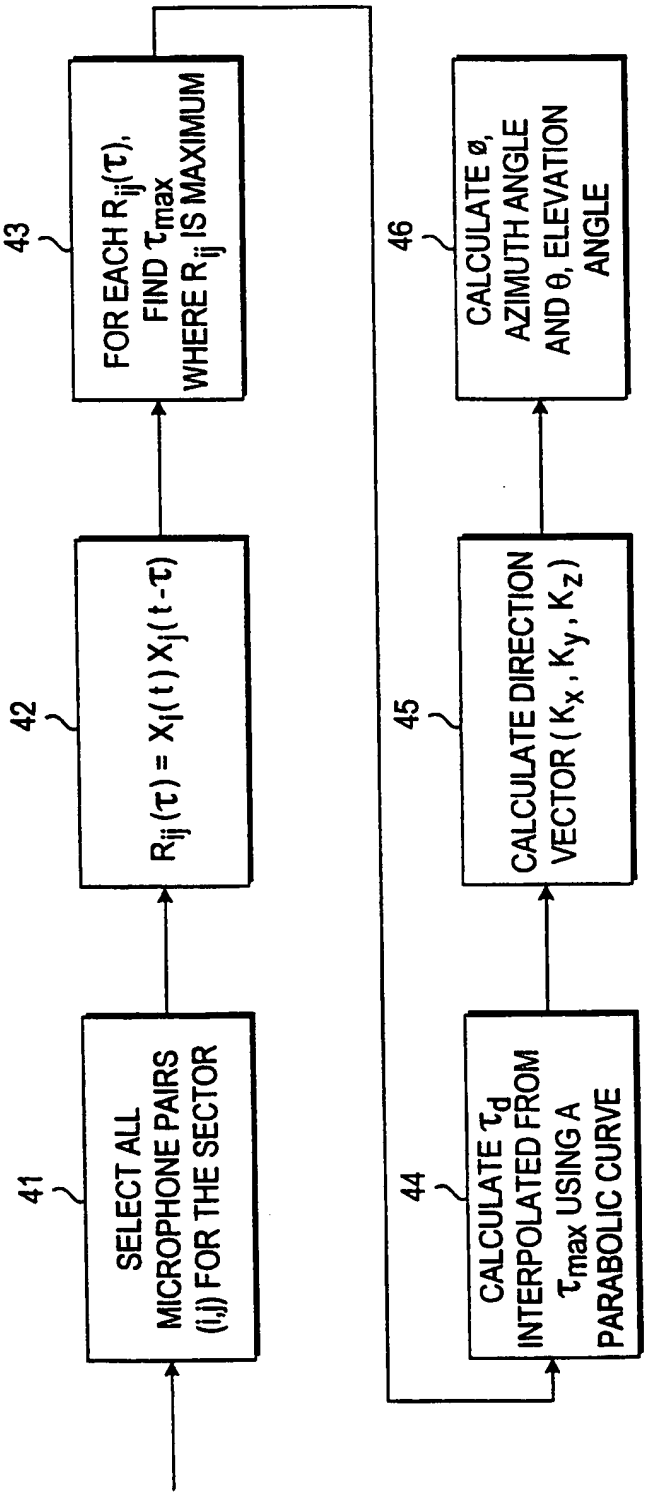
41
SELECT ALL MICROPHONE PAIRS (i,j) FOR THE SECTOR
42
$R_{ij}(\tau) = X_i(t) X_j(t-\tau)$
43
FOR EACH $R_{ij}(\tau)$, FIND τ_{max} WHERE R_{ij} IS MAXIMUM
44
CALCULATE τ_d INTERPOLATED FROM τ_{max} USING A PARABOLIC CURVE
45
CALCULATE DIRECTION VECTOR (K_x, K_y, K_z)
46
CALCULATE ø, AZIMUTH ANGLE AND θ, ELEVATION ANGLE
22

FIG. 5

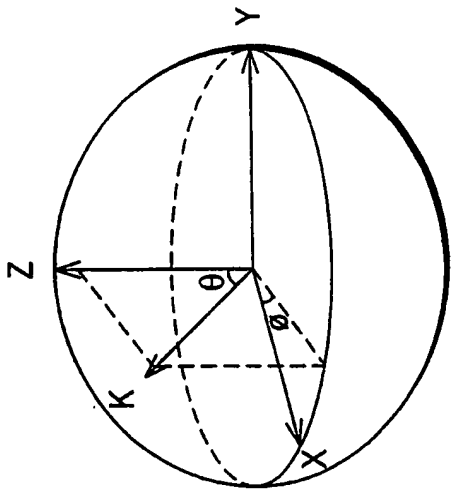

FIG. 6

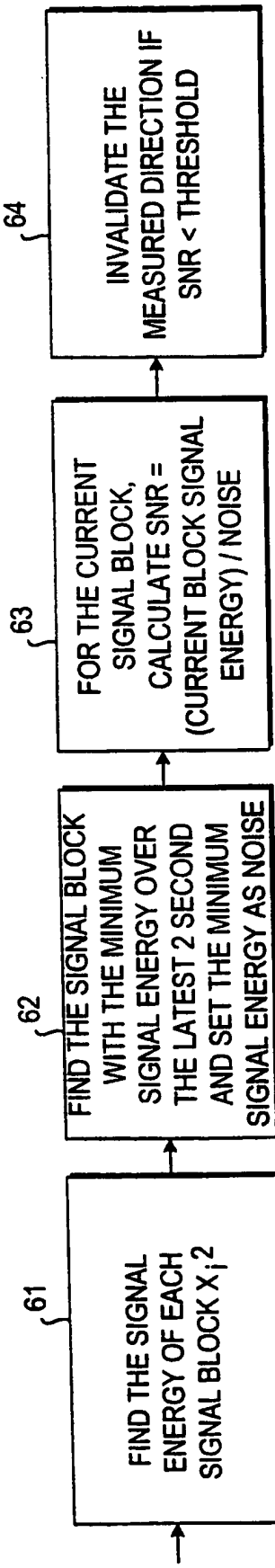
61
FIND THE SIGNAL ENERGY OF EACH SIGNAL BLOCK X_i^2
62
FIND THE SIGNAL BLOCK WITH THE MINIMUM SIGNAL ENERGY OVER THE LATEST 2 SECOND AND SET THE MINIMUM SIGNAL ENERGY AS NOISE
63
FOR THE CURRENT SIGNAL BLOCK, CALCULATE SNR = (CURRENT BLOCK SIGNAL ENERGY) / NOISE
64
INVALIDATE THE MEASURED DIRECTION IF SNR < THRESHOLD

FIG. 7A

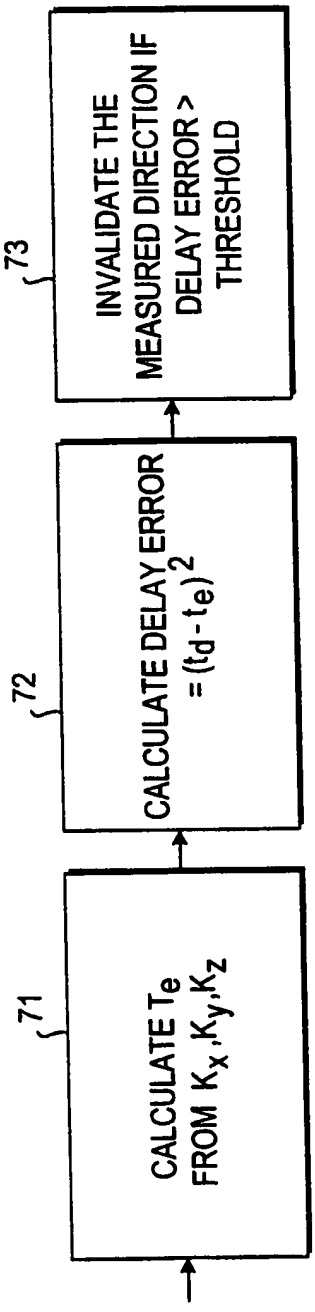
71
CALCULATE T_e FROM K_x, K_y, K_z
72
CALCULATE DELAY ERROR $= (t_d - t_e)^2$
73
INVALIDATE THE MEASURED DIRECTION IF DELAY ERROR > THRESHOLD

FIG. 7B

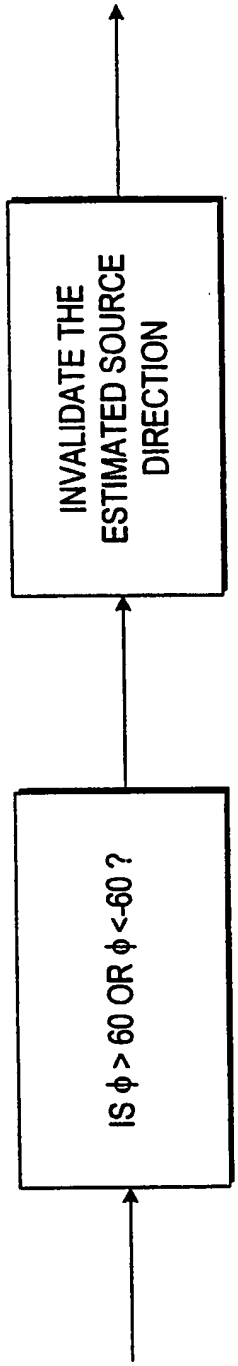
IS ϕ > 60 OR ϕ <-60 ?
INVALIDATE THE ESTIMATED SOURCE DIRECTION

FIG. 7C

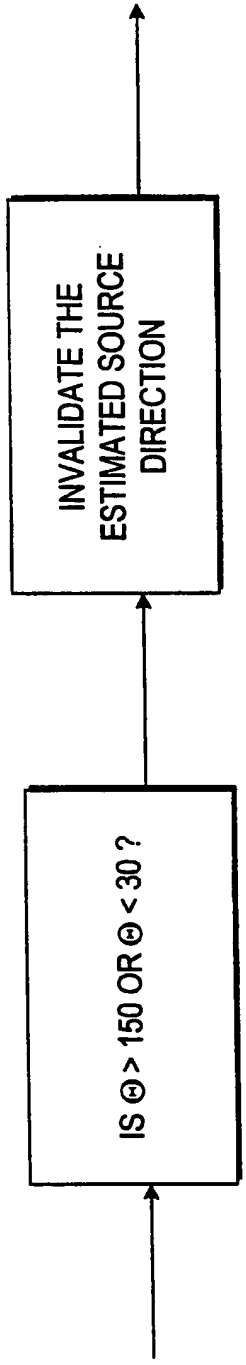
IS Θ > 150 OR Θ < 30 ?
INVALIDATE THE ESTIMATED SOURCE DIRECTION

FIG. 7D

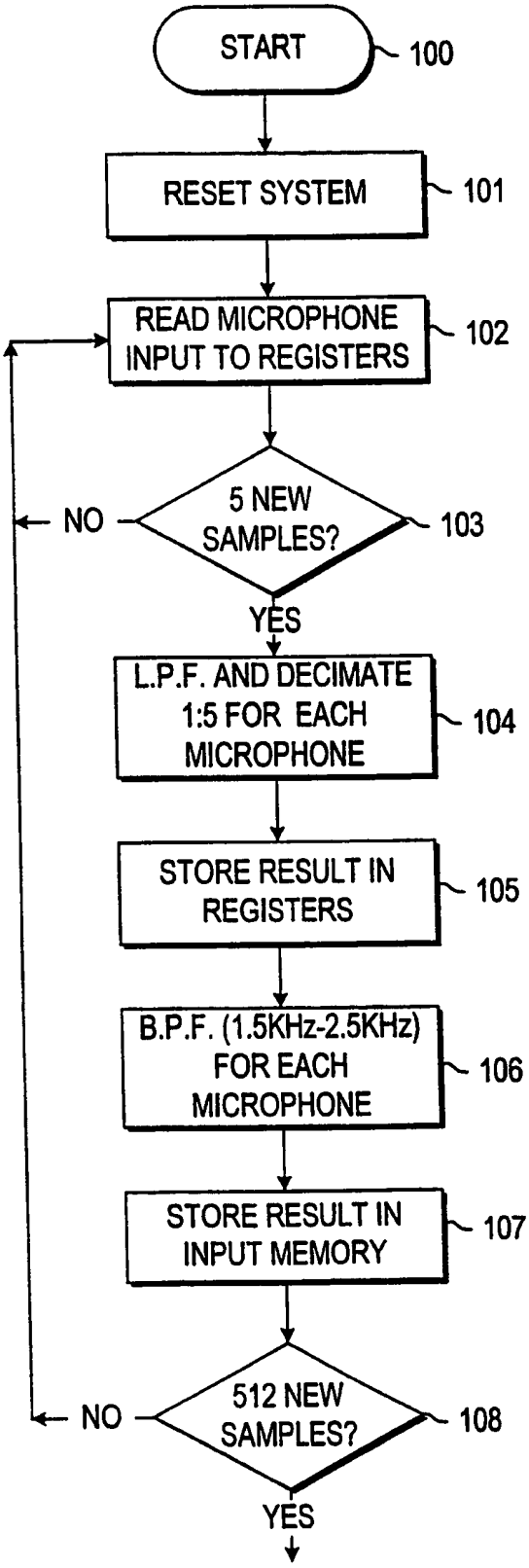
START
100
RESET SYSTEM
101
READ MICROPHONE INPUT TO REGISTERS
102
5 NEW SAMPLES?
103
NO
YES
L.P.F. AND DECIMATE 1:5 FOR EACH MICROPHONE
104
STORE RESULT IN REGISTERS
105
B.P.F. (1.5KHz-2.5KHz) FOR EACH MICROPHONE
106
STORE RESULT IN INPUT MEMORY
107
512 NEW SAMPLES?
108
NO
YES

FIG. 8A

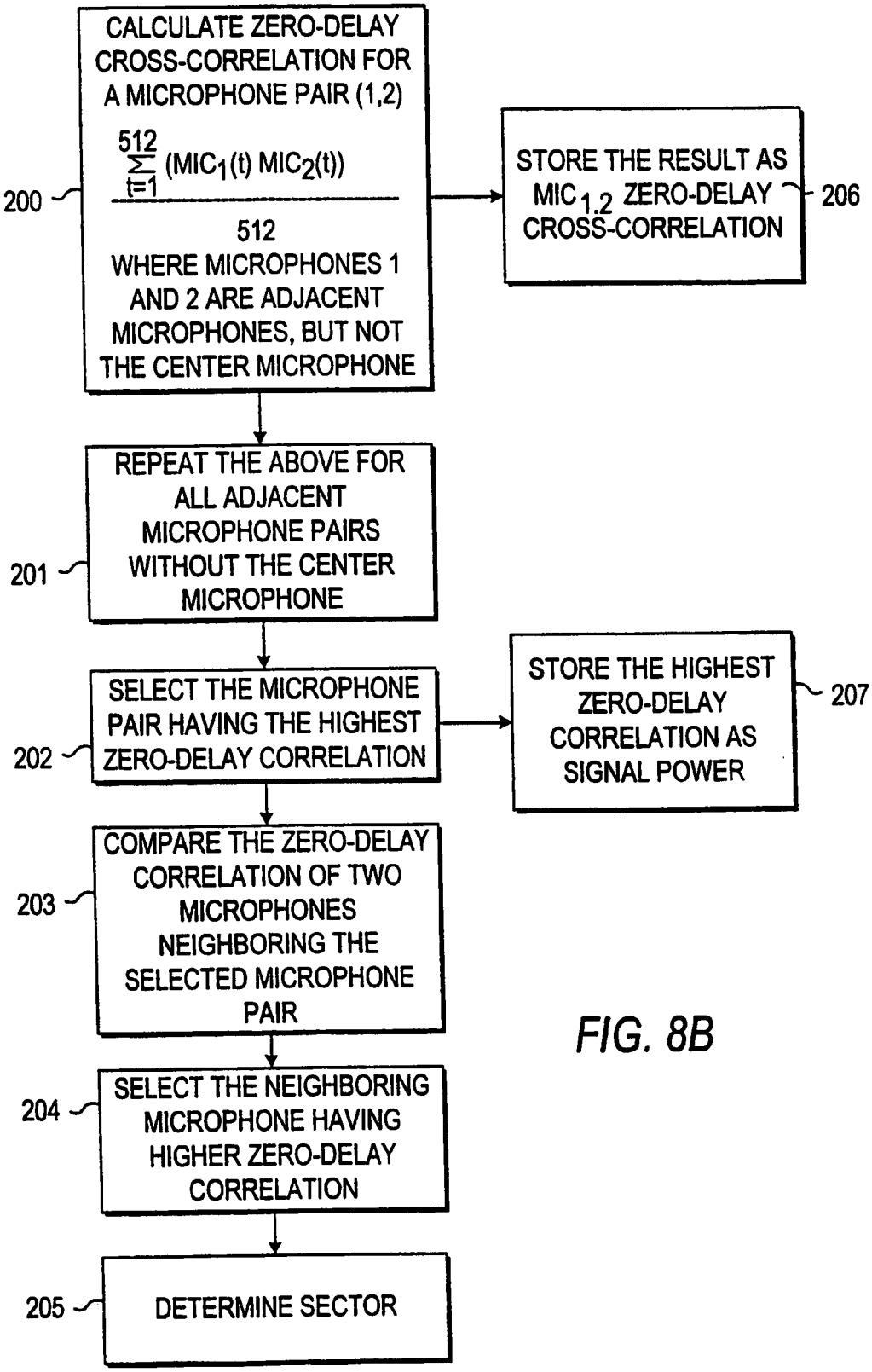
CALCULATE ZERO-DELAY CROSS-CORRELATION FOR A MICROPHONE PAIR (1,2)
$\frac{\sum_{t=1}^{512} (MIC_1(t)\, MIC_2(t))}{512}$
WHERE MICROPHONES 1 AND 2 ARE ADJACENT MICROPHONES, BUT NOT THE CENTER MICROPHONE
200
STORE THE RESULT AS $MIC_{1,2}$ ZERO-DELAY CROSS-CORRELATION
206
REPEAT THE ABOVE FOR ALL ADJACENT MICROPHONE PAIRS WITHOUT THE CENTER MICROPHONE
201
SELECT THE MICROPHONE PAIR HAVING THE HIGHEST ZERO-DELAY CORRELATION
202
STORE THE HIGHEST ZERO-DELAY CORRELATION AS SIGNAL POWER
207
COMPARE THE ZERO-DELAY CORRELATION OF TWO MICROPHONES NEIGHBORING THE SELECTED MICROPHONE PAIR
203
SELECT THE NEIGHBORING MICROPHONE HAVING HIGHER ZERO-DELAY CORRELATION
204
DETERMINE SECTOR
205

FIG. 8B

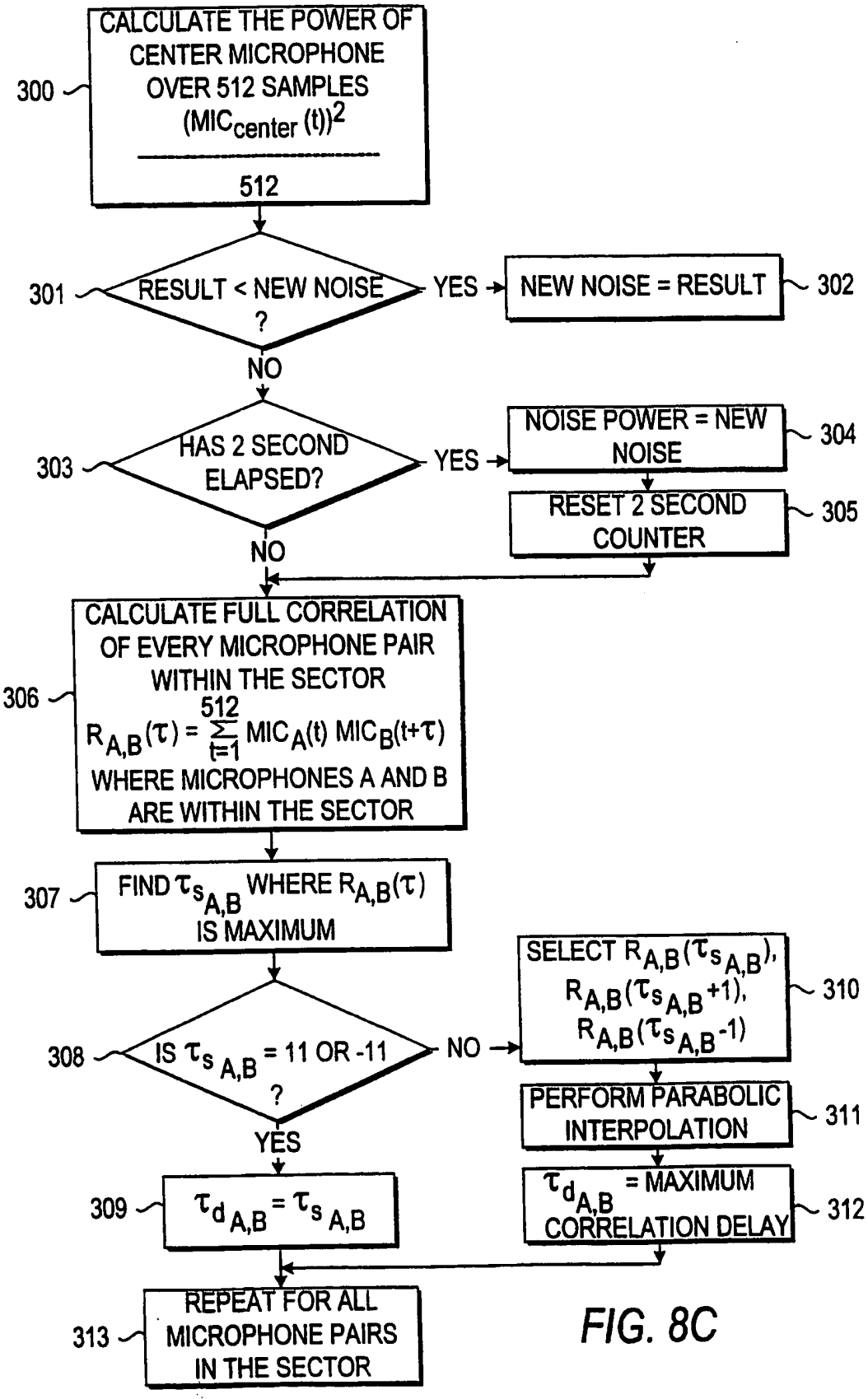
CALCULATE THE POWER OF CENTER MICROPHONE OVER 512 SAMPLES
$(MIC_{center}(t))^2$
512
300
RESULT < NEW NOISE ?
301
YES
NEW NOISE = RESULT
302
NO
HAS 2 SECOND ELAPSED?
303
YES
NOISE POWER = NEW NOISE
304
RESET 2 SECOND COUNTER
305
NO
CALCULATE FULL CORRELATION OF EVERY MICROPHONE PAIR WITHIN THE SECTOR
$R_{A,B}(\tau) = \sum_{t=1}^{512} MIC_A(t)\, MIC_B(t+\tau)$
WHERE MICROPHONES A AND B ARE WITHIN THE SECTOR
306
FIND $\tau s_{A,B}$ WHERE $R_{A,B}(\tau)$ IS MAXIMUM
307
IS $\tau s_{A,B}$ = 11 OR -11 ?
308
NO
SELECT $R_{A,B}(\tau s_{A,B})$, $R_{A,B}(\tau s_{A,B}+1)$, $R_{A,B}(\tau s_{A,B}-1)$
310
PERFORM PARABOLIC INTERPOLATION
311
$\tau d_{A,B}$ = MAXIMUM CORRELATION DELAY
312
YES
$\tau d_{A,B} = \tau s_{A,B}$
309
REPEAT FOR ALL MICROPHONE PAIRS IN THE SECTOR
313

FIG. 8C

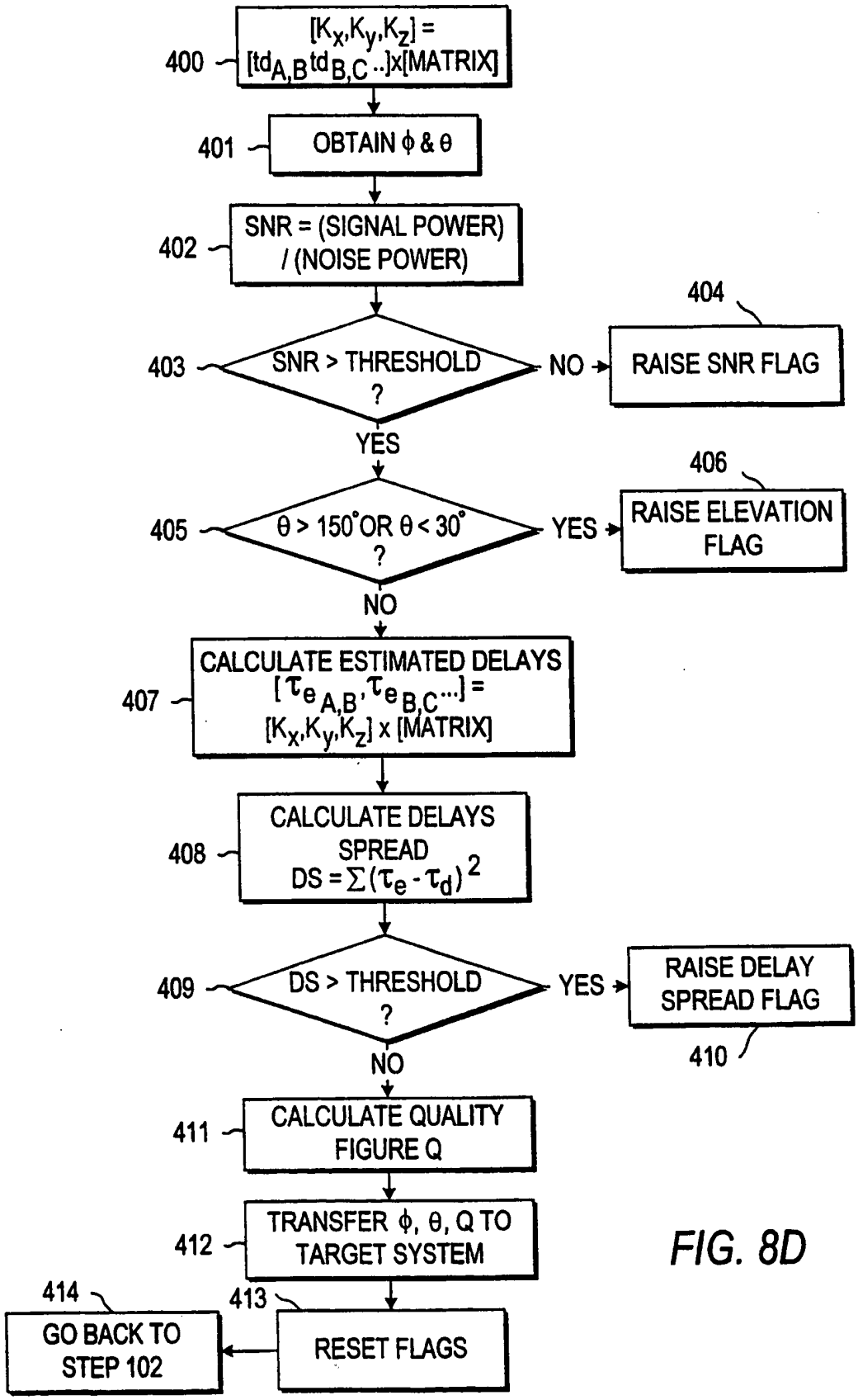
400
[Kx,Ky,Kz] = [tdA,B tdB,C ..]x[MATRIX]
401
OBTAIN φ & θ
402
SNR = (SIGNAL POWER) / (NOISE POWER)
403
SNR > THRESHOLD ?
NO
404
RAISE SNR FLAG
YES
405
θ > 150° OR θ < 30° ?
YES
406
RAISE ELEVATION FLAG
NO
407
CALCULATE ESTIMATED DELAYS [τeA,B, τeB,C ...] = [Kx,Ky,Kz] x [MATRIX]
408
CALCULATE DELAYS SPREAD DS = Σ(τe - τd)2
409
DS > THRESHOLD ?
YES
RAISE DELAY SPREAD FLAG
410
NO
411
CALCULATE QUALITY FIGURE Q
412
TRANSFER φ, θ, Q TO TARGET SYSTEM
413
RESET FLAGS
414
GO BACK TO STEP 102

FIG. 8D

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/08012

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :G01S 5/02, H04B 11/00
US CL :367/125, 701/210, 702/152

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 367/117,125, 701/209,210, 702/152, 324/302, 381/71.1,94.7

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5,724,270 A (POSCH) 03 MARCH 1998 (03/03/98) FIG. 1	1-72
A, P	US 5,798,983 A (KUHN ET AL) 25 AUGUST 1998 (25/08/98), FIG. 5A	1-72
A, P	US 5,874,918 A (CZARNECKI ET AL) 23 FEBRUARY 1999 (23/02/99) COL. 8	1-72
A, E	US 5,914,912 A (YANG) 22 JUNE 1999 (22/06/99) FIG. 3	1-72

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
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"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed		

Date of the actual completion of the international search
19 AUGUST 1999

Date of mailing of the international search report
30 SEP 1999

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